NORTEL NETWORKS™

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Meridian 1

X11 Release 24.0x

Software Feature Guide

Book 3 of 3

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Feature

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BK 3 of 3

Scheduled Access Restrictions

The Scheduled Access Restrictions (SAR) feature allows a customer to define Trunk Group Access Restrictions (TGAR), Class of Service (COS) restrictions, and Network Class of Service (NCOS) restrictions for different hours and days (typically off-hours and off-days). These TGAR, COS, and NCOS restrictions comprise SAR groups. Each customer may define up to 1000 SAR groups, and one of these groups can be assigned to each customer station or route. Up to eight time periods can be defined for each SAR group, and different restrictions may be applied to each time period.

SAR can be overridden on a single call basis for a station or route by using an authorization code or forced charge account. By using the SAR Disable (SADS), SAR Enable (SAEN), SAR Lock (SALK), or SAR Unlock (SAUN) Flexible Feature Codes these restrictions can be changed on a more permanent basis.

SADS returns the set/route to its normal restriction state. SAEN cancels SADS, returning the set to its SAR state. SALK will occur automatically at a predefined period of time or when the Lock command is dialed by the user. Lock restrictions remain in effect until an SAUN or SADS command is entered. The SALK command can be used on a customer basis or SAR group basis, depending on the authcode used.

Typically, the Flexible Feature Codes can be used to do the following:

— extend off-hour restrictions for weekends or holidays (SALK)

— return to the schedule of access restrictions (SAUN)

— extend normal restrictions into the off-hour period for after hour services (SADS)

— cancel this after hour service (SAEN)

— cause off-hour restrictions to start immediately (SALK followed by SAEN), and

— disallow any calls on an Attendant Console (SALK on SAR group containing the attendant(s)).

Customer attendants that are included in SAR groups are placed in Position Busy when an off-hour or off-day period goes into effect. The restricted attendant can only release existing calls or dial the SAR Flexible Feature Codes. New calls cannot be made. Incoming calls will be directed to any other attendants that are not included in SAR groups and that are not in Position Busy.

If the system is placed in Night Service by an attendant, or the system is automatically placed in Night Service because all attendants are in the Position Busy state, incoming calls are routed to the Night DN. Going into Night Service will automatically place attendants who belong to a SAR group into an SAR Locked and Enabled state. These attendants can only release existing calls or dial the SAR Flexible Feature codes; they cannot make new calls when restricted by SAR.

Operating parameters

The definition of authorization codes for SAR decreases the number of authorization codes available for non-SAR use.

SAR does not apply to Direct Inward System Access (DISA) DNs. DISA can be used to manually modify the SAR schedule using an FFC authorization code.

Telephones and trunks assigned to SAR groups have their Class of Service (COS), Trunk Group Access Restriction (TGAR) and Network Class of Service (NCOS) defined by the SAR schedule of their SAR group.

During the periods that a SAR or SAR lock is in effect, the Controlled Class of Service (CCOS) for the station or trunk is overridden.

If a Facility Restriction Level (FRL) is changed in order to be associated with a different NFCR tree, the NCOS using that FRL is affected. Also, different FRLs, and therefore different New Flexible Code Restriction (NFCR) trees, are used at different times according to the NCOS assigned to the SAR group.

Feature interactions

Access Restrictions

The Trunk Access Restriction Group (TARG) defined for each route is not altered by Scheduled Access Restrictions. Access to the route is denied to any telephone or trunk assigned a Trunk Group Access Restriction code that is part of the TARG.

Automatic Redial

The Scheduled Access Restrictions (SAR) on Automatic Redial (ARDL) redialed calls are set when the call is initiated. If restrictions are changed later, the prior restrictions still apply.

Attendant Clearing during Night Service

Attendant Clearing during Night Service should be equipped with Scheduled Access Restriction (SAR) due to the fact that when Night Service is in effect the only operations that may be performed from Attendant Consoles which are members of a SAR group are:

— release any existing calls, or

— dial the one of the following SAR Flexible Feature Codes:

- Scheduled Access Disable (SADS)
- Scheduled Access Enable (SAEN)
- Scheduled Access Lock (SALK), or
- Scheduled Access Unlock (SAUN).

Authorization Code Security Enhancement

Authorization Codes can be used to override SAR restrictions. In addition, Authorization Codes are defined for the specific use of SAR FFCs.

Basic Alternate Route Selection

If SAR is equipped when Basic Alternate Route Selection (BARS) is set up, a NCOS value between 0 and 99 must be defined for each time period.

Call Detail Recording

If configured, Call Detail Recording (CDR) A-type records are printed for SAR Flexible Feature Codes functions.

Charge Account, Forced

Forced Charge Account (FCA) can be used to override Scheduled Access Restrictions (SAR) on a per-call basis, provided the current Class of Service (COS) of the telephone or trunk is CUN, TLD, or CTD. The current COS is the COS in force according to the SAR schedule. If an Authorization Code that sets the COS to CUN, TLD, or CTD is dialed before the FCA, the call is allowed. FCA sets the COS to UNR and the Network COS (NCOS) to the NCOS defined in LD 15, provided that FCA is enabled on both a customer and telephone/trunk basis.

Class of Service

Sets defined in LD 10 and 11, and trunks defined in LD 14 which are assigned a SARG number, have their Class of Service defined by the SAR schedule of their SAR group.

Controlled Class of Service

During normal hours, Controlled Class of Service (CCOS) restrictions override normal telephone restrictions. During off-hour periods or times when a Scheduled Access Restrictions (SAR) LOCK is in effect, however, Scheduled Access Restrictions apply. When the LOCK or off-hour period ends, CCOS restrictions continue to apply until they are removed or SAR becomes effective again. Whether a CCOS controller or electronic lock is used to activate CCOS, there is no indication to the user when Scheduled Access Restrictions are in effect, overriding CCOS restrictions. A telephone defined in LD 10 or 11 or a trunk defined in LD 14, which is assigned an SAR group number, has its Class of Service defined by the SAR schedule of its SAR group.

Coordinated Dialing Plan

If SAR is equipped when Coordinated Dialing Plan (CDP) is set up, a NCOS value between 0 and 99 must be defined for each time period.

Direct Inward System Access

Direct Inward System Access (DISA) numbers are not assigned to SAR groups and therefore are not affected by SAR schedules.

DISA can be used to manually modify the SAR schedule, provided that the correct FFC and Authorization Code are dialed.

Electronic Lock Network Wide/Electronic Lock on Private Lines

The SAR feature overrides Electronic Lock.

Multi-Tenant Service

If a SAR is assigned to a tenant, any set belonging to the tenant will follow this SAR schedule unless the set belongs to a SAR group. The set's Scheduled Access Restrictions override any SAR assigned to the tenant.

Network Alternate Route Selection

If SAR is equipped when Network Alternate Route Selection (NARS) is set up, a NCOS value between 0 and 99 must be defined for each time period.

Network Class of Service

When a Network Class of Services (NCOS) is changed, it may be necessary to alter the NCOS values defined for each SAR roup in LD 88. The NCOS value, which defines the facility restriction level and hence the NFCR trees, is used as defined by the SAR schedule. Also, different FRLs, and hence different NFCR trees, are used at different times according to the NCOS assigned to the SARG.

New Flexible Code Restriction

If a Facility Restriction Level (FRL) is changed to be associated with a different NFCR tree, any NCOS which uses that FRL is affected. In turn, the NCOS assigned to a SAR group may also be affected.

Office Data Administration System

Office Data Administration System (ODAS) can be used to indicate that telephones have been assigned to an SAR group. ODAS must be equipped in order to print members of a SAR group in LD 81.

Position Busy with Call on Hold

If an attendant in a Scheduled Access Restriction group has a call on hold, the attendant is not placed in Position Busy when the group enters an off-hour period.

Speed Call
Network Speed Call

The System Speed Call and Network Speed Call features ignore the Class of Service and TGAR access restrictions in a SAR schedule, using the Class of Service and NCOS defined in the speed call list.

Trunk Group Access Restriction

SAR does not alter the Trunk Group Access Restriction defined per route.

Feature packaging

Scheduled Access Restrictions (SAR) is package 162. The following packages are also required:

— Call Detail Recording (CDR) package 4

— Basic Authorization Code (BAUT) package 25

— Network Class of Service (NCOS) package 32

— Network Authorization Code (NAUT) package 63

— Multi-Tenant Service (TENS) package 86

— To add the capability for manual modification of schedules, Flexible Feature Codes (FFC) package 139 and Basic Authorization Codes (BAUT) package 25 are required.

— If Call Detail Recording is required, Call Detail Recording (CDR) package 4 must be equipped.

— To make Network Class of Service restrictions effective, Network Class of Service (NCOS) package 32 is required.

— For additional billing information, Charge Account for CDR (CHG) package 23, Charge Account/Authorization Code (CAB) package 24, and Forced Charge Account (FCA) package 52 are required.

Feature implementation

LD 88 – Create or modify Scheduled Access Restrictions as follows:

Prompt	Response	Description
REQ	NEW CHG	Create or change existing data block.
TYPE	SAR	Scheduled Access Restrictions.
CUST	0-99 0-31	Customer number. For Option 11C.
SPWD	xxxx	Secure data password (same password as defined for DISA on a per customer basis in LD 15). ***Note:*** Prompt will not appear to a user with an LAO password.
SGRP	0-999	SAR group number.
SCDR	(NO) YES	(Do not) activate CDR for the SAR FFC commands.
OFFP	1-8	Off-hour period number. Off-hour periods may overlap; the period that starts first has priority until that off-hour period is over.
	<CR>	Go to ICR prompt.
- STAR hh mm	hh mm	Start time. The current start time (hours and minutes) is printed individually after the prompt. Respond with the new start time.
	X	Remove value and return to OFFP prompt.
- STOP hh mm	hh mm	Stop time. The current stop time (hours and minutes) is printed individually after the prompt. Respond with the new stop time.
	X	Remove value and return to OFFP prompt.
- DAYS	d ... d	Respond with a new set of days to be used. Maximum of seven entries in the range of 1-7. Day 1 = Sunday, Day 2 = Monday, etc.

- COS		Off-hour period Class of Service.
	(UNR)	Unrestricted
	CTD	Conditionally Toll-Denied
	CUN	Conditionally Unrestricted
	FR1	Fully Restricted Class1
	FR2	Fully Restricted Class 2
	FRE	Fully Restricted
	SRE	Semi-restricted
	TLD	Toll Denied
- TGAR	(0)-15	Trunk Group Access Restriction.
- NCOS	0-99	Network Class of Service.
- ICR	(NO) YES	Incoming Calls are Restricted.
LOCK	(1)-8	The LOCK prompt is used to indicate which off-hour period is to be used as the LOCK period. The default is Period 1.

LD 88 – If the system is in an off-hour or locked period when a print command is issued, an asterisk appears following the restrictions being used. If lock is in effect, an additional asterisk appears following the lock prompt. The print command allows a tenant number to be entered. The status of a tenant SAR group can be printed.

Prompt	Response	Description
REQ	PRT	Print.
TYPE	SAR	Scheduled Access Restrictions.
CUST	0-99 0-31	Customer number. For Option 11C.
SPWD	xxxx	Secure data password.
TEN	1-511	Tenant number.
SGRP	0-999	Prompted only if no tenant number is entered.

LD 88 – With SAR you must configure the Authcode data block not to automatically generate Authcodes as follows:

Prompt	Response	Description
REQ	NEW	New.
TYPE	AUB	Authcode data block.
CUST	0-99 0-31	Customer number. For Option 11C.
SPWD	xxxx	Secure data password (same password as defined for DISA on a per customer basis in LD 15).
ALEN	1-14	Number of digits in authcodes.
ACDR	YES NO	Activate CDR for authcodes (there is no default response).
RANR	0-511 X	RAN route number for authcode last prompt. Enter X for no entry.
CLAS	(0)-115	Classcode value assigned to authcode.
AUTO	NO	Do not automatically generate Authcodes. ***Note:*** Prompted when NAUT package 63 is equipped and REQ = NEW. The Authcode length must be a minimum of four digits.

LD 88 – Define SAR entries in the Authcode entries data block as follows:

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	AUT	Authcode entries data block.
CUST	0-99 0-31	Customer number. For Option 11C.
SPWD	xxxx	Secure data password (same password as defined for DISA on a per customer basis in LD 15).
CODE	xxx..x	Authcode (1-14 digits).
SARC	YES NO	Allow or disallow Authcode to be used as the SAR authorization code.

- SERV		SAR service functions for SARC (the SERV prompt appears if SARC = YES)
	(END) ENA (LKD) LKA (DSD) DSA (UND) UNA	Enable (Denied) Allowed. Lock (Denied) Allowed. Disable (Denied) Allowed. Unlock (Denied) Allowed ***Note:*** Up to four entries can be made at once.
- SRGP	0-999 ALL	Number of SAR group to be defined or changed. Change all SAR groups.
CLAS	(0)-115	Class code value assigned to authcode. Cycle continues with CODE. When type = AUT, enter X to configure the authcode as an exempt code. When this data is printed, the month the authcode was deactivated is output. The default is 0 when adding authcode entries.
	X	Exempt authcode.

LD 10 – A SAR group can be assigned to individual analog (500/2500 type) telephones by entering the SAR group number in response to the SGRT prompt. If the SAR group number has not been previously defined in LD 88, an error message is issued.

Prompt	**Response**	**Description**
REQ	CHG	Change.
TYPE	500	Telephone type.
...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

LD 11 – A SAR group can be assigned to individual Meridian 1 proprietary telephones by entering the SAR group number in response to the SGRT prompt. If the SAR group number has not been previously defined in LD 88, an error message is issued.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. Terminal Number for the Option 11.
...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

LD 12 – A SAR group can be assigned to individual Attendant Consoles by entering the SAR group number in response to the SGRT prompt. If the SAR group number has not been previously defined in LD 88, an error message is issued. Attendant Consoles do not follow the SAR restrictions defined by the SARG group, but they can be locked by using SAR FFCs.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	1250 2250 ATT	Attendant Console type.
TN	l s c u c u	Terminal Number. Terminal Number for the Option 11.
...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

LD 16 – A SAR group can be assigned to individual trunk route by entering the SAR group number in response to the SGRT prompt. If the SAR group number has not been previously defined in LD 88, an error message is issued.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	RDB	Route Data Block.
CUST	0-99 0-31	Customer number. For Option 11C.
...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

LD 57 – Define Flexible Feature Codes for the SAR disable, SAR enable, SAR lock, and SAR unlock functions.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	FFC	Flexibly Feature Codes data block.
CUST	0-99 0-31	Customer number. For Option 11C.
...		
CODE	aaaa	Specific Flexible Feature Code Type. To change a specific Flexible Feature Code, enter the associated mnemonic then carriage return <CR>. The mnemonic will then be prompted and the Flexible Feature Code can be entered. The Flexible Feature Code may be up to four digits or up to seven digits if DNXP package 150 is equipped.
- SADS	xxxx	SAR Disable code.
- SAEN	xxxx	SAR Enable code.

- SALK	xxxx	SAR Lock code.
- SAUN	xxxx	SAR Unlock code.

LD 93 – A SARG can be entered for each tenant. Sets follow the restriction of this group if they are a member of the tenant and do not belong to a SARG themselves. Respond to the TEN prompt with the tenant number. Respond to the SGRP prompt with the number of the SAR group to be assigned to the tenant.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	TGEN	Tenant SAR data block.
CUST	0-99 0-31	Customer number. For Option 11C.
...		
TEN	1-511	Tenant number.
...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

Feature operation

Modification of SAR Restrictions

SAR restrictions can be modified on a per call basis by using an Authorization code, if the Basic Authorization Code (BAUT) package 25 is equipped.

Also, if the Authorization Code and Flexible Feature Codes packages are equipped, the off-hour periods can be shortened or extended by using the four SAR FFCs.

The Authorization Code feature can be used to allow a user to override a Scheduled Access Restriction on a single-call basis by dialing an Authorization Code (Authcode). Each Authcode is assigned a Class of Service, Trunk Group Access Restriction, and a Network Class of Service. The restrictions associated with the dialed Authcode, apply to the call being made. Thus, by using an Authcode, any facility to which access is allowed depending on the restrictions associated with an Authcode, can be accessed by dialing the set, even though the set may normally be denied access.

Single-Call Modification

The Scheduled Access Restrictions feature does not modify using Authcodes to allow calls to be made on restricted sets. Dial either "SPRE + 6" or the AUTH FFC plus the Authcode associated with the desired restrictions. Once dial tone is returned, indicating a valid code, the call may be dialed as normal.

Off-Hour Period Modification

The SADS, SAEN, SALK, and SAUN FFCs defined in LD 57 can be used to modify off-hour period restrictions, by simply dialing the FFC plus an appropriate Authcode. The Authcode determines if the requested function is allowed and whether the action is to take place on a SAR group or a customer basis. An FFC plus and Authcode for a specific SARG is only accepted from a station within that group, or from a station within a tenant which uses that SAR group.

Entering a Flexible Feature Code plus an Authcode results in the following:

— SALK + Authcode = extend off-hour restrictions for weekends or holidays

— SAUN + Authcode = return to the schedule of access restrictions

- SADS + Authcode = extend normal restrictions into the off-hour period for after hour services
- SAEN + Authcode = cancel this after hour service
- SALK followed by SAEN + Authcode = cause off-hour restrictions to start immediately, and
- SALK on SAR group containing the attendant(s) + Authcode = disallow any calls on an Attendant Console.

Introduced in X11 Release:	24
Networking:	No

Message Waiting Indicator by Directory Number

The Message Waiting Indicator by Directory Number (MWDN) feature increases the flexibility in presenting a message waiting indication on the M2006, M2008, M2016, M2216, and M2616 Meridian 1 proprietary sets. The MWDN feature provides the following functionalities:

— presentation of multiple message waiting indications on one set

— presentation of multiple message waiting indications for one mailbox on more than one set

— presentation of remote message waiting indications for message monitoring

— support for one mailbox for multiple Directory Numbers (DNs)

Multiple message waiting indications on one set

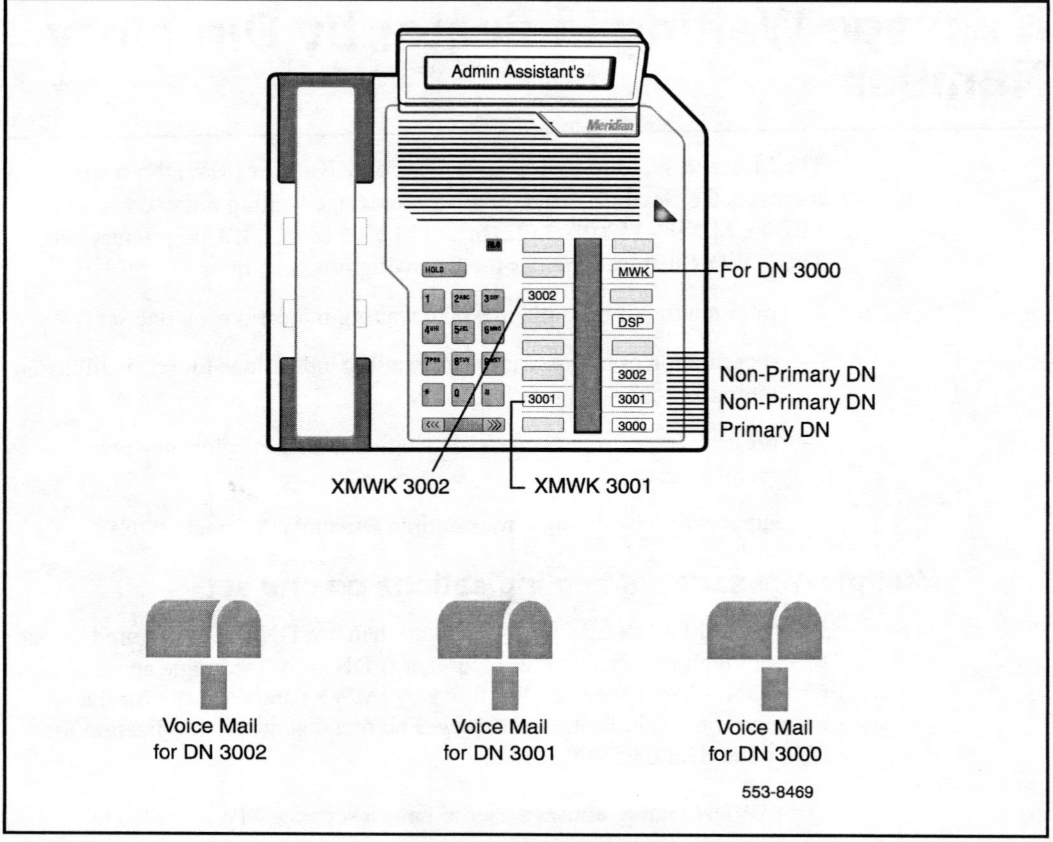
Prior to the MWDN feature, where more than one DN was configured on one set, only the Primary Directory Number (PDN) -- or the single appearance non-PDN -- had a Message Waiting Key (MWK) and the LED for the message waiting indication. There was no message waiting indication for DNs other than the PDN.

The MWDN feature allows a user to have a separate MWK, called the Extended Message Waiting Key (XMWK), for each of the mailbox DNs configured on that set. The DN associated with the XMWK must be configured as a non-PDN on that set.

The XMWK starts flashing when a new voice message is received for the DN associated with this key. Once all the new voice messages have been retrieved, the indication on the XMWK associated with that DN is canceled.

Multiple message waiting indications on one set has application for environments where one set has the DNs for several individuals. Figure 58 shows a scenario where an administrative assistant monitors the DNs for several individuals from his/her set.

Figure 58
Multiple message waiting indications on one set



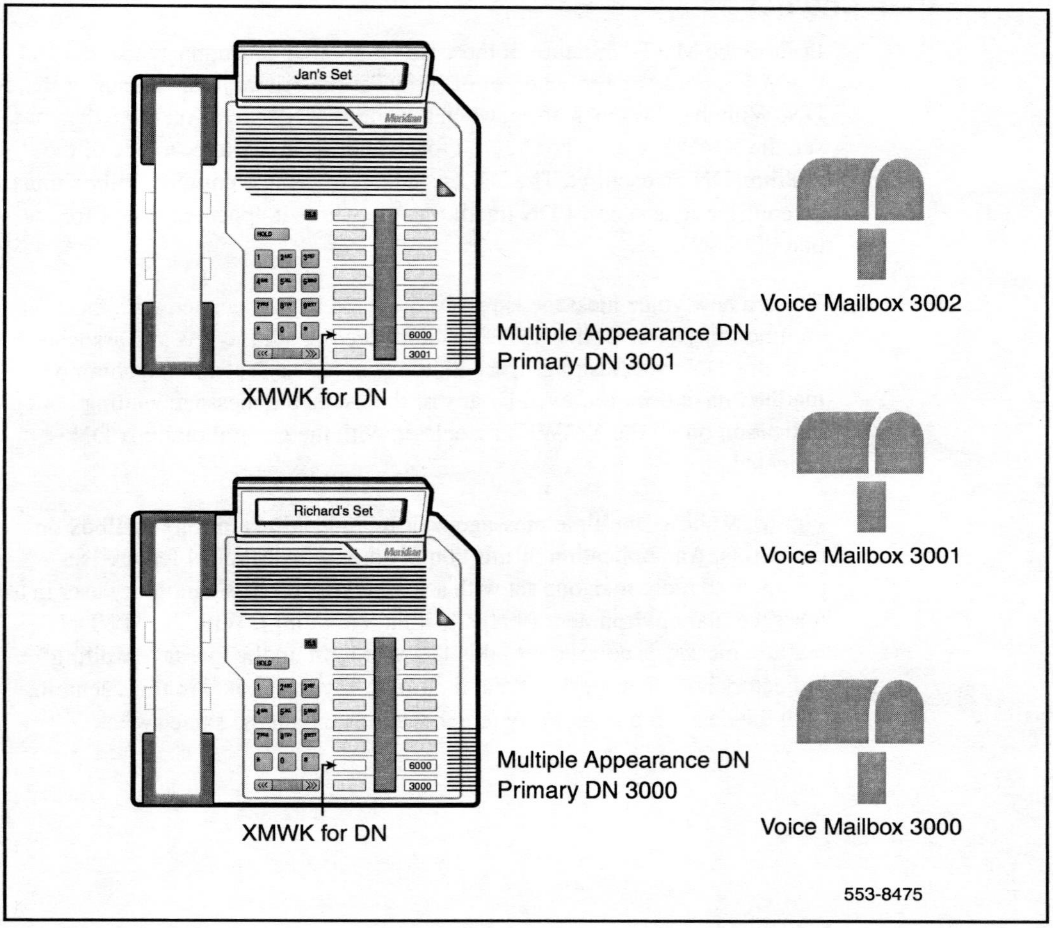
Multiple message waiting indications for one mailbox on more than one set

Prior to the MWDN feature, if there was more than one appearance of a DN, the MWK could be turned on or off only for the primary appearance of that DN. With the MWDN feature, when a mailbox DN appears on more than one set, the XMWK can be configured for the non-primary appearance of the mailbox DN on each set. The DN associated with the common mailbox must be configured as a non-PDN on all the sets where it appears (except for the one PDN set).

When a new voice message is received for the DN associated with the common mailbox, all the XMWKs configured on all the sets and associated with this DN start flashing. Once all the new messages from the common mailbox have been retrieved by any of the users, the message waiting indication on all the XMWKs associated with the general mailbox DN is canceled.

Figure 59 shows multiple message waiting indications for one mailbox on many sets. An application of this component of the MWDN feature is a person with more than one set with a shared DN, such as a mobility user in a macrocelular environment (that is, within a building). With the MWDN feature, messages coming into this DN will light up the message waiting indicators both their desk set and their mobility telephone. In this scenario, both the desk set and mobility telephone must be on the same switch.

Figure 59
Multiple message waiting indications for one mailbox on more than one set



Remote message waiting indication for message monitoring

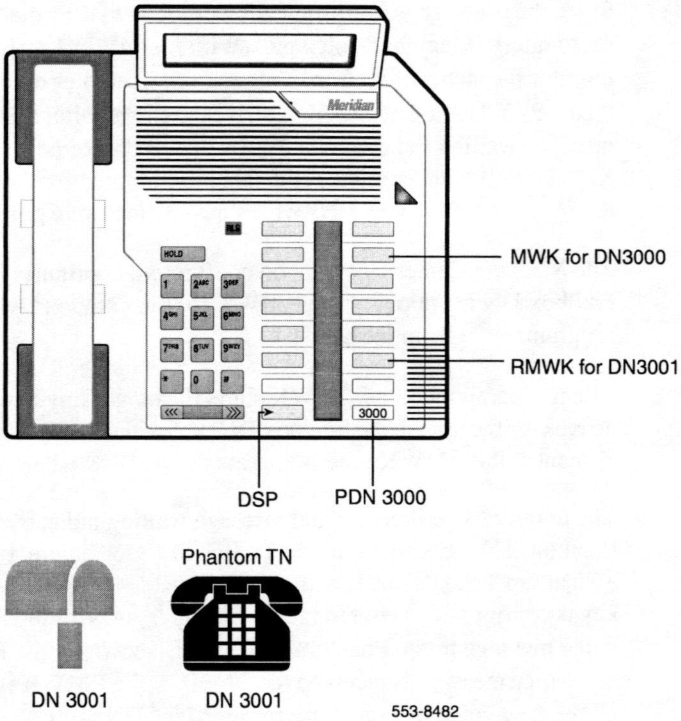
Prior to the MWDN feature, if a new voice message was received, users had to see the message waiting indication or log in to their mailbox from a remote set to query if they had voicemail. With the MWDN feature, users can monitor the status of their mailboxes from a remote set without logging into their sets.When a new message arrives to the monitored mailbox DN, the message waiting indication is propagated to the Remote Message Waiting Key (RMWK) on a remote set that is programmed for that mailbox DN. The RMWK monitors those DNs which have at least one primary appearance.

The Message Center DN must be configured; configuring the monitored mailbox DN is optional. The RMWK for the mailbox DN is user programmable from the set.

When programmed, the RMWK starts flashing if any new voice message arrives for the associated mailbox DN; if not, the RMWK remains steadily lit. To cancel the RMWK function, press the RMWK when it is lit or flashing.

The temporary redirection and message waiting indicator propagation of a Phantom TN uses this component of the MWDN feature. Figure 60 illustrates a Phantom TN, DN 3001, with a RMMA/RMMO class of service. A RWMK key is configured on a set to monitor the messages for the DN 3001. Any new voice message to the Phantom DN 3001 is shown on the RMWK. When a new voice message is received for DN 3001, the RMWK starts flashing; once all the new voice messages are retrieved for DN 3001, the RMWK becomes steady lit.

Figure 60
RMWK operation when a Phantom TN is call forwarded to a set

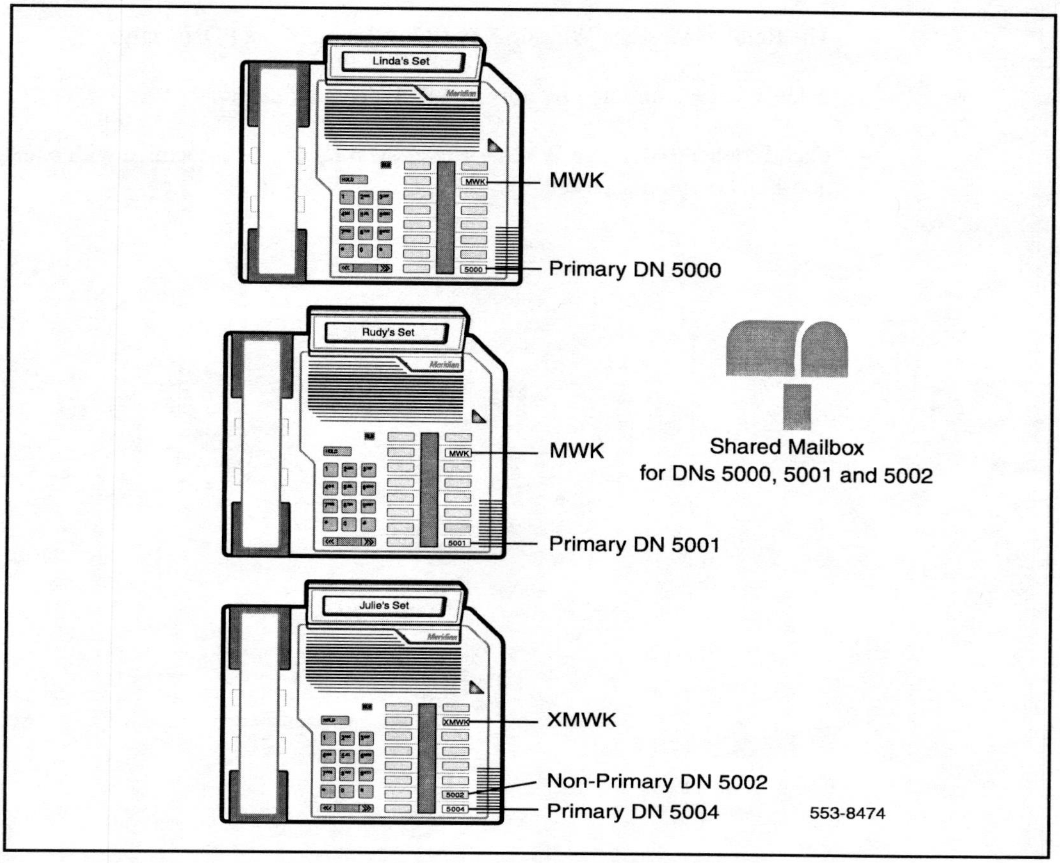


One mailbox for multiple DNs

Prior to the MWDN feature, three DNs could be associated with one mailbox; however, only the PDN which shares the mailbox displayed the message waiting indication. The MWDN feature extends the message waiting indication to all Meridian 1 proprietary set appearances on which the three DNs sharing the mailbox are configured.

This feature is used in an environment such as a technical support area with up to three technicians having their own DN but sharing a common mailbox. Figure 61 shows DNs 5000, 5001 and 5002 with a shared mailbox.

Figure 61
One mailbox for multiple DNs



Operating parameters

MWDN supports features within the same node; it does not support features on different nodes across a network. For example, MWDN supports Meridian Mail if it is on the same node; the MWDN feature does not support Meridian Customer Defined Network (MCDN) messaging services across a network.

Meridian Mail 9 is required to support one mailbox for multiple DN functionality. The Voice Mailbox Administration (VMBA) package 246 must be equipped to enable the functionality of one mailbox for multiple DNs.

The MWDN feature does not support message waiting indication in the macrocellular environment.

The Remote Message Waiting Key (RMWK) monitors PDNs only.

A DN can be monitored by only one RMWK at a time.

Each Extended Message Waiting Key (XMWK) can be associated with one non-PDN only on each set.

Feature interactions

Display key

With the MWDN feature, the Display key (DSP) shows the Message Center DN and the mailbox DN associated with the XMWK and the RMWK. This display occurs when a user presses the DSP and then either the XMWK or the RMWK. If there is no mailbox DN associated with the RMWK, only the Message Center DN is displayed.

Multiple Appearance DN

For the Multiple Appearance DN feature:

— On sets where the DN is configured as a PDN, the message waiting indication occurs on the MWK and red LED.

— On sets where the DN is configured as a non-PDN, the message waiting indication occurs on the XMWK and the red LED depending upon the LMPN or LMPX class of service. The LMPN class of service is defined as the red LED reflects the status of the mailbox associated with the PDN. The LMPX class of service is defined as the red LED reflects the status of the mailboxes associated with both PDN and non-PDNs.

The RMWK can be used to monitor a Multiple Appearance DN if the DN has at least one primary appearance.

Phantom Terminal Number

The Phantom Terminal Number feature permits users to define and configure Terminal Numbers (TNs) with no associated physical hardware. The Phantom TN can be associated temporarily with a physical set. With the MWDN feature, a user can monitor the mailbox associated with the Phantom DN through the RWMK on a Meridian 1 proprietary set.

Feature packaging

The MWDN feature requires these packages:

— Digit Display (DDSP) package 19

— Message Waiting Lamp Maintenance (MWC) package 46

— Voice Mailbox Administration (VMBA) package 246

Feature implementation

For all the following tasks, first enable the Message Center in the Customer Data Block in LD 15. See page 1929.

To configure multiple message waiting indications on one set or on many sets:

— Configure the class of service options for the Extended Message Waiting Key (XMWK) and its LCDs on Meridian 1 proprietary sets in LD 11. See page 1930.

To configure remote message waiting indications on one set:

— Configure the Remote Message Waiting Key (RMWK) to monitor remote sets with a mailbox in LD 11. See page 1931.

— Configure new class of service options to enable analog (500/2500) sets to be monitored remotely in LD 10. See page 1932.

Note: All sets with primary DNs to be monitored **must** be configured as RMMA/RMMO.

— Configure new class of service options to enable Meridian 1 proprietary sets to be monitored remotely in LD 11. See page 1933.

Note: All sets with primary DNs to be monitored **must** be configured as RMMA/RMMO.

To configure message waiting indications on sets where DNs sharing a mailbox appear:

— Configure the class of service option to enable the message waiting indication for all the analog (500/2500) sets on which the DNs sharing the mailbox are configured in LD 10. See page 1934.

— Extend the message waiting indication function to all the Meridian 1 proprietary sets on which the DNs sharing the mailbox are configured in LD 11. See page 1935.

LD 15 – Enable the Message Center in the Customer Data Block.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	RDR	Call Redirection.
CUST	xx	Customer number. xx = 0-99 for Options 51C-81C. xx = 0-31 for Option 11C.
OPT		Options.
	MCI	Message Center Included..

LD 11 – Configure the class of service option for the Extended Message Waiting Key (XMWK) and its LEDs on the Meridian 1 proprietary sets.

To configure the XMWK key, the DN to be associated with the XMWK must be configured as a non-Primary Directory Number (non-PDN) on this set.

> ***Note:*** The DN associated with the XMWK must not have an XMWK already associated with it on this set.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	aaaa	Type of set. aaaa = 2006, 2008, 2016, 2216, 2616.
...		
CLS		Class of service option.
	MWA	Message Waiting Allowed. (MWD) = Message Waiting Denied.
	LMPX	The red LED on the Meridian 1 proprietary sets reflects the status of the mailbox associated with both the PDNs and non-PDNs with the associated Extended Message Waiting Keys (XMWKs) or the Remote Message Waiting Keys (RMWKs). (LMPN) = The red LED on Meridian 1 proprietary sets reflect the status of the mailbox associated with the PDNs.
...		
KEY		Telephone function key assignments.
	xx XMWK xxxx yyyy	Extended Message Waiting indication key Where: xx = key number xxxx = Message Center DN yyyy = mailbox DN ***Note:*** XWMK cannot be configured on key 0.

LD 11 – Configure the Remote Message Waiting Key (RMWK) to monitor remote sets with a mailbox.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	aaaa	Type of set. aaaa = 2006, 2008, 2016, 2216, 2616.
...		
CLS		Class of service option.
	MWA	Message Waiting Allowed. (MWD) = Message Waiting Denied.
	LMPX	The red LED on the Meridian 1 proprietary sets reflects the status of the mailbox associated with both the PDNs and non-PDNs with the associated Extended Message Waiting Keys (XMWKs) or the Remote Message Waiting Keys (RMWKs). (LMPN) = The red LED on Meridian 1 proprietary sets reflect the status of the mailbox associated with the PDNs.
...		
KEY		Telephone function key assignments.
	xx RMWK xxxx [yyyy]	Remote Message Waiting indication key Where: xx = key number xxxx = Message Center DN [yyyy] = DN to be monitored {optional}

LD 10 – Configure new class of service options to enable analog (500/2500) sets to be monitored remotely.

> ***Note:*** All sets with primary DNs to be monitored **must** be configured as RMMA/RMMO.

Prompt	**Response**	**Description**
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	500	Analog (500/2500) set.
TN	l s c u cu	Terminal Number: l = loop, s = shelf, c = card, u = unit for Options 51-81C. c = card, u = unit for Option 11 C.
...		
CLS		Class of service option.
	MWA	Message Waiting Allowed. MWD = Message Waiting Denied.
	RMMA RMMO	Allow the set to be remotely monitored for messages. Allow the set to be remotely monitored for messages and allow the set to override, if it is being monitored already. (RMMD) = Deny set for Remote Monitoring of Messages.

LD 11 – Configure the class of service to enable Meridian 1 proprietary sets to be monitored remotely.

> ***Note:*** All sets with primary DNs to be monitored **must** be configured as RMMA/RMMO.

Prompt	**Response**	**Description**
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	aaaa	Type of set. aaaa = 2006, 2008, 2016, 2216, 2616.
TN	l s c u cu	Terminal Number: l = loop, s = shelf, c = card, u = unit for Options 51-81C. c = card, u = unit for Option 11 C.
CLS		Class of service option.
	MWA	Message Waiting Allowed. MWD = Message Waiting Denied.
	LMPX	Enable the red LED on the supported Meridian 1 proprietary sets to reflect the status of the mailboxes associated with both PDN and non-PDNs. (LMPN) = do not enable the red LED on the supported Meridian 1 proprietary sets to reflect the status of the mailboxes associated with both PDN and non-PDNs.
	RMMA RMMO	Allow set for Remote Monitoring of Messages. Allow set for Remote Monitoring of Messages and Override, if it is being monitored already. (RMMD) = Deny set for Remote Monitoring of Messages.

LD 10 – Extend the message waiting indication function to all the analog (500/2500) sets on which the DNs sharing the mailbox are configured.

> ***Note:*** Voice Mailbox Administration (VMBA) must be configured before configuring one mailbox supporting Multiple Appearance DNs. Refer to *Message Center description and operation* (553-2691-100) for information on configuring VMBA.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	500	Analog (500/2500) set.
...		
CLS		Class of service option.
	MWA	Message Waiting Allowed. MWD = Message Waiting Denied.
	SMWA	Allow Extended Message Waiting Indication. (SMWD) = Deny Extended Message Waiting Indication.

LD 11 – Extend the message waiting indication function to all the Meridian 1 proprietary sets on which the DNs sharing the mailbox are configured (whether the DNs are PDN or non-PDN).

> ***Note:*** Voice Mailbox Administration (VMBA) must be configured before configuring one mailbox supporting Multiple Appearance DNs. Refer to *Message Center description and operation* (553-2691-100) for information on configuring VMBA.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	aaaa	Type of set. aaaa = 2006, 2008, 2016, 2216, 2616.
...		
CLS		Class of service option.
	MWA	Message Waiting Allowed.
	SMWA	Allow Extended Message Waiting Indication. (SMWD) = Deny Extended Message Waiting Indication.

Feature operation

Remote message monitoring:

1 Press the RMWK with the set in idle position.

2 The set winks and displays RMWK XXXX (where XXXX is the existing mailbox DN) prompting for a new mailbox DN. If there is no mailbox DN, the set displays RMWK.

3 Enter the new mailbox DN.

4 The screen displays the digits. Press the RMWK to validate the mailbox DN.

Note: If you press the RMWK without entering the digits, the RMWK remains programmed for the DN which was stored previously. If there is no DN stored, the Overflow tone is given.

5 If the DN is invalid, the Overflow tone is given. If the mailbox DN is valid:

— If the set on which this DN is configured as PDN has a class of service set to RMMA or RMMO and is not monitored:

- the RMWK starts flashing if there are any new voice messages for this DN.
- the RMWK lamp becomes steady lit and the screen changes to idle mode if no new voice message exists for this DN.

— If the set on which this DN is configured as PDN has a class of service set to RMMO and is being monitored by another set, this set overrides and continues to monitor.

— Overflow tone is given if any set on which this DN is configured as PDN has class of service set to RMMD.

— Overflow tone is given if any set on which this DN is configured as a PDN has class of service set to RMMA and is being monitored by another set.

6 To cancel remote message monitoring, press the RMWK when it is lit or flashing.

Introduced in X11 Release:	15
Networking:	No

Message Waiting Lamp Maintenance

This maintenance enhancement alleviates the "dark effect" when neon lights are tested in low ambient light conditions.

Because the dark effect is inherent to neon lamps, it is recommended that PBXT Message Waiting Lamp tests not be run during low ambient light conditions. The line card detector circuitry can register lamp failures under these circumstances, and the Message Waiting Lamp test may be unreliable. Lamps are listed as faulty when they fail the test once in three attempts.

The PBXT Message Waiting Lamp tests can be run under one of the following conditions:

— automatically at a system-specified time, or

— manually at any time (LD 32).

Automatic scheduling should consider low traffic times, when there is still enough ambient light to avoid the dark effect. To prevent the automatic scheduling of LD 32, LD 32 must be excluded from the daily routines ("midnights") and the system-defined hour must be the default "X" value.

When the hour defined defaults to the "X" value, an error message is output to remind the customer that the PBXT tests are still part of the daily routines, unless LD 32 is removed from the list.

Operating parameters

There are no feature requirements.

Feature interactions

There are no interactions with other features.

Feature packaging

The Message Waiting Lamp Maintenance feature requires Message Waiting Center (MWC) package 46.

Feature implementation

LD 17 – Define the time for the maintenance tests.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN OVLY	Configuration Record. Release 19 gate opener.
OVLY	(NO), YES	Change overlay area options.
- PBXH	hh	PBX Hour for maintenance tests, where: hh = hour for tests, 0-23.
	x	Enter x if no tests are to be performed.

Feature operation

No specific operating procedures are required to use this feature.

Message Waiting Unconditional

This feature enhances the use of the Message Indication key (MIK) and Message Cancellation key (MCK) by an Automatic Call Distribution (ACD) message center agent or message center attendant.

This feature enhancement applies to a Network Message Center. It is configured on a customer basis in LD 15.

Operating parameters

There are no feature requirements.

Feature interactions

ACD Message Center

The operation of ACD Message Center telephones is basically the same as an ACD system with incoming call queues and available agent queues. The ACD Message Center cannot operate in combination with an Attendant Message Center. However, if all telephones are in the Make Busy mode (not logged in), Message Center calls can be routed to the attendants who can then function as the message center. Queue overflow features are allowed for a Message Center ACD DN in the same way as for any other ACD system with the properly equipped package. Other ACD features, such as RAN and Music, operate as for a normal ACD system with the appropriate packages.

A Message Center operator cannot originate calls on the MSG IN-CALLS key; therefore originating features are not applicable on this key. Separate DN keys must be provided for these functions.

DN Message Center

The Message Center DN must be the prime DN, otherwise all normal features can be assigned to this DN.

Attendant Message Center

Once a call is extended to an ACD Message Center by an attendant, it is released completely from attendant operation, and features, such as recall and camp-on, cannot be activated. For calls extended to a DN Message Center, normal attendant functions, such as recall and camp-on, can be used. Other attendant functions operate normally.

Call Forward (All Calls)

Call Forward should be denied at telephones serving as the message center. On a telephone basis, Call Forward takes precedence over the message center. If a call is forwarded to another telephone, activation of message waiting depends on whether or not the second telephone has message waiting allowed.

If the system is equipped with X11 Release 19 or later, Call Forward Message Waiting dialtone can be provided to 500/2500 type telephones. This is an indication that Call Forward All Calls is active and a message is waiting at the message center.

Call Forward, Internal Calls

The Message Center treats Internal CFW in the same way as Call Forward All Calls (CFAC).

If the system is equipped with X11 release 19 or later, Call Forward Message Waiting dialtone can be provided to 500/2500 type telephones as an indication that Call Forward, Internal Calls is active and a message is waiting at the message center.

Call Forward Busy

Call Forward Busy (CFB) should be denied at telephones serving as the message center. An option is provided to allow DID calls to a busy telephone to be routed to the message center. If this option is selected by the customer, message waiting takes precedence over the customer-defined path for CFB.

Call Forward No Answer

Call Forward No Answer (CFNA) should be denied at telephones serving as the message center. On a telephone user basis, message waiting takes precedence over the customer defined path of CFNA.

The capability to light and extinguish message waiting lamps can be used in conjunction with CFNA to simulate a multiple message center. Any telephone equipped with message lamps, but without message waiting allowed class of service, can CFNA to specified DNs on the telephones equipped with MSG INDIC and MSG CANC key/lamp pairs.

These telephones have the capability to light or extinguish message waiting lamps by manually entering the DN of the telephone for which a message was taken. Call processing is the normal call processing for CFNA, not the message center call processing. When a call is forwarded, the MSG INDIC lamp does not light since this is not true message center operation.

Call Transfer/Conference from an Analog (500/2500 type) telephone

Message waiting interrupted dial tone is not provided when the user flashes the switchback to activate Call Transfer or Conference. The normal dial tone for this purpose is provided.

Flexible Call Forward No Answer to any DN

Flexible Call Forward No Answer (CFNA) to any DN forwards unanswered calls to a pre-designated CFNA DN. All telephones with message waiting allowed have the CFNA DN assigned to the message center regardless of whether Flexible CFNA has been selected by the customer or whether CFNA is allowed or denied for the telephone.

Hunting

Hunting should be denied at telephones serving as the message center. On a user basis, hunting takes precedence over message waiting. However, message waiting can be activated after hunting provided the hunted telephone is message waiting allowed and does not answer the call. If desired, the MC DN can be specified as the hunt number.

Listed Directory Number

A message center can be assigned to a Listed Directory Number (LDN) and behaves in a similar manner to an attendant message center. The calls come in on an LDN ICI instead of the MSG CENTER ICI, and direct message calls do not activate the MSG CANC key. The operator must access the user telephone directly to cancel that telephone's message indication.

Ring Again for an Analog (500/2500 type) telephones

Message waiting interrupted dial tone is not provided when the user flashes the switch back to activate Ring Again. The normal dial tone for this purpose is provided.

User Selectable Call Redirection

User Selectable Call Redirection allows the user to perform two tasks:

- To assign the four redirection DNs from the telephone. These DNs include the CFNA DN and the external CFNA DN (if it exists).
- To change the way the number of ringing cycles are defined for Flexible Call Forward No Answer (CFNA). One of three options can now be selected from the telephone.

This feature does not support Basic Rate Interface (BRI) telephones.

Feature packaging

Message Waiting Center (MWC) package 46.

Feature implementation

LD 15 – Configure the Message Waiting Unconditional feature enhancement for a customer.

Prompt	Response	Description
...		
OPT	(MWUD) MWUA	Message Waiting Unconditional feature enhancement (denied) allowed.

Feature operation

The current operation is such that, if an internal call or an incoming external call to a station is not answered, the caller may leave a message at the message center (ACD agent or message center attendant). To activate or deactivate a message waiting indication on the desired station, the ACD agent or attendant presses the Message Indication key (MIK) and Message Cancellation key (MCK), respectively. To use this method when the message center has an active call, the active call must be placed on hold, or the message center attendant has to be placed in position busy or the ACD agent in Not Ready state before the MIK/MCK may be activated.

The enhanced operation allows the Message Indication key (MIK) and Message Cancellation key (MCK) to be used unconditionally (i.e., if there is a call presented to the message center, and not answered, pressing the MIK or MCK takes precedence over the presented call).

> *Note:* This enhancement applies only to presented calls which have not been answered. If the message center has a call already established, the current operation applies.

Introduced in X11 Release:	All
Networking:	Yes

Multifrequency Compelled Signaling

Multifrequency Compelled (MFC) signaling is a signaling protocol that allows a Meridian 1 to exchange information with another Meridian 1, or a Central Office (CO)/Public Service Telephone Network (PSTN). In addition to providing a medium for transmitting called address digits, MFC offers both exchanges an extensive set of signals describing the status and category of the calling and called parties.

Information on MFC signaling and MFC signaling related features can be found in the *Multifrequency Compelled Signaling* (553-2861-100) NTP.

Introduced in X11 Release:	20.0x
Networking:	No

Multi-language Messaging

Options 51C, 61C and 81 software has a system of message reporting that issues reports containing English sentences in addition to error codes and data (introduced in X11 Release 18), and the Error Message Lookup feature (introduced in X11 Release 19) that enables Northern Telecom Publication (NTP) explanations of any error code to be displayed on the TTY. The Multi-language Messaging feature enhances these capabilities by providing an additional language besides English. In addition, the capacity to toggle from one language to another without suspending system operations is provided.

Operations, Administration and Maintenance users on Options 61C and 81 will now have the ability to have some messages currently printed in English to be displayed and logged using another language. The following messages are affected:

— Maintenance and start-up messages specific to Options 61C and 81, with the following exceptions:

- Messages printed by the VxWorks OS
- Read Only Memory Firmware (ROM F/W) messages
- Interactive messages from overlays

— Explanation texts printed by the System Message Lookup Utility (MLU).

Operating parameters

This feature applies to the Meridian 1 Options 51C, 61C and 81.

Every system can support only one language besides English, and changing this language to another language is only possible after re-initializing the system.

Upgrading the messages database to a more recent version is only possible by upgrading the software.

The language is selected for the whole system and changes simultaneously on all configured terminals and log files.

Messages are logged in the Report Processing Tool (RPT) log file as they come (that is, in the language currently configured in LD 17). Hence, the log file may contain messages in both English and the alternate language.

After the language option has been changed in LD 17, some messages may be displayed in the previous language, because they were sent to the printer queue immediately before service change.

At system start, the first messages will be displayed in English, since the current language will not yet have been read from the disk.

It is not possible for the current feature to enable translation of interactive or hard-coded messages in the Meridian 1 system.

Translation is not possible for the following messages:

— Options 61C and 81 installation tools screens

— LDs 135 and 137 (interactive messages)

— Application modules

— New/existing tools for database consistency checks

— VxWorks OS

— Messages printed during initialization and SYSLOAD by the Meridian 1 software

— Liquid Crystal Display (LCD) displays on the Central Processing Unit (CPU) board, and

— ROM messages.

No new hardware is required for this feature.

Feature interactions

This feature is an improvement based on the Message Lookup Utility (MLU) and the Report Processing Tool (RPT).

Feature packaging

In Release 20, Latin American Spanish (MLMS_SPL) package 279 is defined.

The following package must be activated for the Multi-language Messaging feature to operate: Multi-language TTY Input/Output (MLIO) package 211.

The following package must be activated to gain access to the System Message Lookup feature: System Message Lookup (SYS_MSG_LKUP) package 245.

Feature implementation

LD 17 – One new prompt has been added to select which messages are to be printed. The new TRNS prompt is issued only if one of the language packages is equipped.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PARM	Release 19 gate opener.
...		
PARM	YES	System parameters.
...		
- NDIS	(NO) YES	New distinctive ringing.
- TRNS		Selects which messages are going to be translated.
	(NONE)	NONE = Help and Options 61C and 81 specific system messages are printed in English.
	HELP	HELP = Help is printed in the translated language and Options 61C and 81 specific system messages are printed in English.
	BOTH	BOTH = Help and Options 61C and 81 specific system messages are printed in the translated language.
		Note: The translated language printed is dependent on the software packaging.

Feature operation

No specific operating procedures are required to use this feature.

Multi-Party Operations

Multi-Party Operations (MPO) introduces a number of capabilities. The capabilities are:

Call Join

Allows Meridian 1 proprietary telephone users to conference a held party into an active call without having to redial the held party.

Call Join applies to all Meridian 1 proprietary telephones, regardless of the Class of Service assigned, that are equipped with a Three-party (AO3) or Six-party (AO6) Conference key and at least one secondary DN or Call Waiting key.

This feature allows a Meridian 1 proprietary telephone user to conference a party held with an active party on their set, or transfer the active party to the held party by forming a conference then disconnecting.

Call Join is not available at the Attendant Console.

Three-party Service

Allows analog (500/2500 type) telephone users to toggle between two parties with the option of forming a conference between them, or releasing the active party and reconnecting the held party. Included under the Three-party Service capability are:

— **Three-party Service Timer** – A programmable timer to allow dialing of a Control Digit after a Register Recall.

— **Consultation Call Disconnect Option** – An option to provide alternative treatment to the parties involved in a Consultation call when the Consultation connection is released.

Three-party Service applies to analog (500/2500 type) telephones with Three-party Service Allowed (TSA) Class of Service.

During a normal two-party call, the user can place the established call on hold and originate another call. After the second call is established, the user can:

a Dial the Conference Control Digit (CNFD) to form a three-party conference between the user, held, and active parties, or transfer the active party to the held party by forming a three-party conference then disconnecting.

b Dial the Toggle Control Digit (TGLD) to exchange the active and held calls.

c Dial the Disconnect Control Digit (DISD) to release the active call and reconnect the held call.

Programmable Control Digits

The Control Digits may be programmed in LD 15.

Three-party Service time out treatment

A timer is provided on a customer basis to activate an optional time out treatment. The optional time out treatment is to release the active party and connect to the held party if the controlling party of a Consultation call does not dial a Control Digit within the time specified during a Register Recall. The result is the same as if the controlling party had dialed the Control Digit assigned to the DISD function.

The optional time out treatment is selected by the user by responding to the Control Digit Time Out (CDTO) prompt in LD 15 with a value in the range of 2 to 14 seconds.

If the user selects either the default, 14 seconds, or enters 14 seconds, then the operation is as it was prior to the introduction of the Three-party Service time out treatment. That is, if a Control Digit is not entered within 14 seconds, Overflow Tone is provided for 14 seconds, after which the call returns to its previous state; the held party remains on hold and the consulted party is reconnected. If a Register Recall is performed while Overflow Tone is given the call returns to its previous state.

Six-party Conference Enhancement for analog (500/2500 type) telephones

Provides analog (500/2500 type) telephone users with the ability to Conference up to six parties.

If the MPO package is equipped, then Six-party Conference Enhancement is available to analog (500/2500 type) telephones with a combination of the TSA and existing C6A Classes of Service.

This capability is an extension of Three-party Service which allows the user to build a conference of up to six parties by consulting and selectively adding members through the use of Control Digits.

Ignore Switchhook Flash

Provides the ability, on a customer basis, to ignore a Switchhook Flash from analog (500/2500 type) telephones. This eliminates the confusion between a flash signal and a dial "1" signal on Dial Impulse analog (500/2500 type) telephones, especially when the Dial Impulse analog (500/2500 type) telephones have been assigned DTN Class of Service.

If the flash is to be ignored, analog (500/2500 type) telephones must have a Ground (EARTH) Button in order to use features which require a Register Recall.

Forced Register Recall

Provides an option, on a customer basis, to force analog (500/2500 type) telephone users assigned DTN Class of Service to issue a Register Recall before dialing a Control Digit.

If the system does not have the Forced Register Recall option activated, then a Switchhook Flash is interpreted as a dial "1", default CNFD, causing that Control Digit assignment to be activated.

Manual return after enquiry (Manual Hold)

Provides an option (MHLD) to require analog (500/2500 type) telephone users to issue a Register Recall to return to a held party following a Consultation dialing time out.

At present, when an analog (500/2500 type) telephone places a party on hold by using a Register Recall, sets with DIP Class of Service receive 30 seconds, while Digitone (DTN) Class of Service sets receive 14 seconds, of Special Dial Tone followed by 14 seconds of Overflow Tone before the held party is reconnected. During this period the held party is listening to silence, or RAN if equipped.

The Manual Return after Enquiry option (MHLD) controls the way the held party is reconnected to the controlling party. If MHLD = NO (the default), the controlling party is automatically reconnected to the held party after the overflow tone timeout. If MHLD = YES, the controlling party receives silence indefinitely after the overflow tone timeout until a second recall is performed to retrieve the held party. There is no automatic reconnection of the held party. The controlling party may manually return to the held party by performing a second Register Recall during Special Dial Tone, Overflow Tone or during the silence period.

Recovery of Mis-operation during Call Transfer

The Recovery of Mis-operation during Call Transfer feature provides protection against having calls lost due to mis-operation of the Call Transfer feature. Mis-operation occurs whenever the user initiates an unexpected action that would normally cause a call to be lost.

If a station user tries to perform an illegal Call Transfer (for example, Call Transfer to a vacant number or Call Transfer to a busy extension), the station user receives the appropriate indication on the Consultation connection (for example, Overflow Tone and Busy Tone). However, since transfer in the ringing state is allowed, the user may still mis-operate and complete the Call Transfer operation immediately after dialing the desired number.

If a Meridian 1 proprietary telephone user attempts to complete a Call Transfer by pressing the Call Transfer key, the call is only transferred if the dialed party is in the ringing state or in the Consultation state with the controlling party. In other states the attempt to Call Transfer is ignored.

When an analog (500/2500 type) (500/2500) set initiates a supervised call transfer to a DN in any other state than ringing, the call transfer mis-operation treatment is dependent upon the option chosen for AOCS (all other cases) in the customer data block (LD 15). If any set (either analog or digital) attempts a blind transfer in ringing state, the misoperation treatment is dependent upon the option chosen for Ring No Answer (RGNA) type of mis-operation assigned in LD 15.

A number of options are available, where a call is transferred while the transferred station is ringing. For example if AAR, or DAR is selected, the transferred station will ring for an optional number of ring cycles (RCY2). On the expiration of this timer, the transferring set is rung back for an optional number of ring cycles (RCY1) with an optional recall ringing cadence. If the transferring station does not answer during the optional ringing cycles (RCY1), the transferred call will be forwarded to the attendant or Night Service DN (AAR) if external or disconnected (DAR) if internal.

The Recovery options are specified for both RGNA and AOCS cases in LD 15 when the MPO package is equipped. Separate treatment can be specified for external and internal calls.

Switchhook Contact Bounce

The situation occurs when an analog (500/2500 type) telephone goes on-hook. Switchhook contact bounce during disconnect may be interpreted by the system as a switchhook flash followed by an on-hook. When this occurs there is an unintended Call Transfer to the attendant or other type of mis-operation.

In order to resolve this problem, with the MPO package equipped, the software is modified to delay recognition of any action for a minimum of 256 milliseconds following receiving a valid switchhook flash from analog (500/2500 type) telephones. During this delay, any signaling received from the parties involved is ignored.

Operating parameters

Tones and cadences are limited by their availability on the equipped Tone and Digit Switch (TDS) card on pre-Meridian 1 systems.

For enhanced functionality of the Multi-Party Operations, the following features should be equipped:

— Automatic Hold for Meridian 1 proprietary telephones

— Ground Button and Flash timers, and

— Recall of mis-operation ringing cadence and Control and Special Dial Tones requires the Flexible Tones and Cadences (FTC) feature.

Feature interactions

Access to Paging trunks

Analog (500/2500 type) telephones with TSA Class of Service are restricted from initiating a Consultation connection while connected to a paging trunk.

Access to Recorded Dictation trunks

Analog (500/2500 type) telephones with TSA Class of Service are restricted from initiating a Consultation connection while connected to a dictation trunk.

Attendant Administration

Attendant Administration allows certain station classes of service to be altered. The operation of Attendant Administration is modified so that if an attendant tries to alter either XFA or XFD Class of Service, then Three-party Service (TSA) Class of Service is disallowed. The TSA and XFA Classes of Service are mutually exclusive. When XFA is assigned, TSA will be disallowed if it was not configured. XFD is not mutually exclusive with TSA, but TSA will not be automatically assigned if the Class of Service is changed to XFD. TSA Class of Service cannot be assigned via Attendant Administration.

This feature can not be used to setup the Three-party Service TSA Class of Service.

Attendant Break-In

Break-In is not allowed to the party receiving the patience tone or the misoperation ringback.

Break-In with Secrecy

For Multi-Party Operation (MPO), the operation of features, such as going on-hook and releasing from a call, during the BKIS conference between the attendant and the desired party, takes precedence over MPO operations for those cases where the treatment differs from that defined by the customer.

All network nodes must have MPO software, with identical Multiple-party Operation (MPO) options. Otherwise, MPO options in the desired party's node have precedence.

Pertaining to MPO options, if the undesired party is not located on the same node as the desired party, the undesired party is considered as an external party on the desired party node.

Attendant Forward No Answer

Multi-Party Operations – Recovery of Mis-operation During Call Transfer takes precedence over NFNA and NFNS for DID/DOD/CO calls.

When a DID/DOD/CO call is transferred from one station to another station on the same node, Ring Again No Answer has priority over NFNA and NFNS.

Attendant Recall

For analog (500/2500 type) telephones with TSA Class of Service, Attendant Recall is accomplished by performing a Register Recall during the two-party connection and dialing the Attendant DN.

Attendant Recall with Splitting

The Multi-Party Operations (MPO) feature introduces a new Class of Service, Three Parties Service Allowed (TSA), for analog (500/2500 type) telephones. It allows certain keys on these sets to be programmed for conference, toggle between sets, and disconnect. However, the toggle function will be disabled if a call is transferred to the attendant because of the Attendant Recall with Splitting feature.

Call Forward All Calls

A set which has activated Call Forward All Calls can still initiate calls and become the controlling party of a consultation connection. In this case, if the set mis-operates, then Multi-Party operations, while re-ringing the controlling party as a part of mis-operation recovery, ignores the Call Forward All Calls indication present on the controlling party.

Call Forward No Answer

For Call Transfer with Ring No Answer (RGNA) if the user has selected an option other than Standard, the optional treatment has priority over the CFNA option selected in the LD 15. If the user has chosen the standard option for RGNA, the call will be treated as a normal CFNA call, and handled according to the options selected for CFNA in LD 15. Once the call is routed to a Night DN during recovery of mis-operation and the Night DN does not answer, the call is treated according to the NFNA and FDN options chosen for the Night DN. The Night DN can use flexible CFNA DN in two levels. MPO mis-operation does not change the operation of the DNFD timer if one has been configured in LD 15.

Call Pickup

Analog (500/2500 type) telephones with PUA and TSA Class of Service can pick up a call only if they are not involved in another call. After picking up a call, the user can form a Consultation connection and dial Programmable Control Digits as normal.

Call Pickup, Directed

Users of analog (500/2500 type) telephones involved in a Three-Party Service call cannot pick up another call by dialing the SPRE code.

Call Transfer

Analog (500/2500 type) telephones with TSA Class of Service perform a supervised Call Transfer by going on-hook after establishing a conference. This differs from operation with XFA Class of service, where transfer can be achieved by going on-hook during Consultation connection. If an analog (500/2500 type) telephone with TSA Class of Service goes on-hook during consultation connection, it is treated as mis-operation of All Other Cases and the recovery actions are done based on the CCDO and AOCS options selected in LD 15. If CDOC = NO, an analog (500/2500 type) telephone can achieve a transfer by going on-hook after establishing a conference.

During the Consultation connection, the non-controlling parties are restricted from using Call Transfer, Conference, and Three-party Service features.

Call Waiting

An analog (500/2500 type) telephone may be assigned both CWA and TSA Classes of Service. The user can establish a Consultation connection by answering Call Waiting during an active established call. If this is done, Control Digit features (CNFD, TGLD, and DISD) are available. Note that Programmable Control Digit TGLD, rather than a switchhook flash, is used to toggle the calls. Operation with XFA Class of Service is unchanged.

The Three-party Service feature changes the operation of Call Waiting for all analog (500/2500 type) telephones as follows (regardless of whether the sets have TSA Class of Service. If an analog (500/2500 type) telephone user activates Waiting during an active call so as to establish a Consultation connection, and if the user goes on-hook during the Consultation connection, the operation is treated as an AOCS mis-operation. This recovery of mis-operation will take place even if the MPO package is not equipped. In this case, the controlling party will be re-rung by the held party regardless of the CCDO and the recovery of mis-operation options.

If an analog (500/2500 type) telephone user attempts to set up a Consultation connection by dialing a busy DN and if the Call Waiting conditions are satisfied, the controlling party will hear ringback tone and the active party will hear Call Waiting tone. If the controlling party goes on-hook before the active party has answered, the held call is disconnected regardless of the MPO options and Call Waiting tone is removed from the active party.

Call Waiting Redirection

Recovery on Misoperation of Call Transfer – Call Transfer with Ring No Answer (RGNA)

With the Call Waiting Redirection feature enabled, if the Controlling Party goes on-hook to complete the call transfer before the Active Party answers the Call Waiting call and before the CFNA timer applied Call Waiting Redirection feature times out, there is no change.

With the Call Waiting Redirection feature enabled, if the CFNA timer applied by the Call Waiting Redirection feature times out before the Call Transfer completes in the Ring No Answer (RGNA) state, CFNA treatment is given by the Call Waiting Redirection feature only if the RGNA option is defined to be Standard (that is, operation as it was prior to the introduction of the Multi-Party Operations feature).

For Call Transfer with Ring No Answer, if the user has selected an option other than Standard treatment, the RGNA option selected has priority over the CFNA option selected in the Customer Data Block. With the Call Waiting Redirection feature enabled, the non-Standard RGNA option will also be enforced. There are no interactions in the functioning of Multi-Party Operations for the Attendant After Recall, Disconnect After Recall, Attendant After Recall, Overflow, and Disconnect RGNA call treatment options.

As the transferred set tries to re-ring the transferring set, if the transferring set is busy, call redirection will again try Call Forward All Calls, Hunting, and Call Waiting in that order. Call Waiting Redirection will not apply CFNA treatment to the unanswered Call Waiting call as the non-Standard RGNA option selected has priority over the CFNA option selected in the Customer Data Block, and thus have priority over Call Waiting Redirection CFNA treatment.

Recovery on Misoperation of Call Transfer – Misoperation of Call Transfer for All Other Cases

This type of misoperation occurs when the transferring party attempts to complete the transfer in several other non-RGNA scenarios. There is no interaction with these Multi-Party Operations scenarios and the Call Waiting Redirection feature.

Camp-on

Camp-on to a controlling party DN which is involved in a Consultation connection is not permitted. However, Camp-on is allowed at non-controlling parties DN's which are involved in the Consultation connection.

Camp-On, Forced Override

With Multi-Party Operations (MPO), when a consultation call is made on a set equipped with Priority Override, a control digit has to be dialed from the set to perform a recall and return the call on hold.

China – Supervised Analog Lines

As in the cases with Call Transfer and Conference, the call type of the first active call determines whether battery reversal or hook flash supervision applies. Also, supervision signaling is not supported for the second call. A disconnect supervision signal is extended only when the last party disconnects.

China – Toll Call Loss Plan

When a user toggles between one party and another, the Toll Loss Plan is inserted on the active call if it is a toll call. If the user toggles to a non-toll call, the Toll Loss Plan is removed.

Conference

Current Conference feature for analog (500/2500 type) telephones with C6A is not affected by conference with TSA Class of Service.

The Call Join feature allows a user of a Meridian 1, Meridian 1000 series, or digital telephone to conference in or transfer a third party to a party held on the user's telephone, without having to dial the third party. The user can then hang up.

The patience tone or the Misoperation ringback is not applied to a conference party

Display of Calling Party Denied

When three parties are joined using the Call Join capabilities of the Multi Party Operations feature, display information is not provided on any of the conferee's sets. When setting up a conference call, by conferencing one set at a time, the display on the conferee's set is in accordance with the individual set's Class of Service. If one set leaves a three party conference, display information on the remaining sets is based on the individual Class of Service of each set.

End-to-End Signaling

The party receiving the patience tone or the Misoperation ringback is not able to use End-to-End Signaling.

Enhanced Music on Hold

Analog (500/2500 type) telephones with TSA Class of Service can receive music when put on hold during Three-party Service.

Enhanced Night Service

Enhanced Night Service allows a mis-operated call involving a Direct Inward Dial (DID) trunk to queue at the Night Service DN.

Group Hunt

As per the existing Multi-Party Operations (MPO) feature, recovery of misoperation of call transfer will not be applied to incoming calls which are transferred on ringing to a Pilot DN by transferring parties who are waiting in GPHT queues for service.

Last Number Redial

For analog (500/2500 type) telephones with TSA Class of Service, the first call of a Consultation connection is stored as the last number. Last Number Redial (LNR) is possible whenever Dial Tone or Special Dial Tone is given.

Night Service

If the system is in Night Service mode, mishandled calls which are routed to the attendant are rerouted to the appropriate Night Service DN. External trunk calls, other than DID, are queued till they are answered.

TIE trunk calls are not queued at the Night Service DN. If the Night Service DN is busy, TIE calls are disconnected.

Off-hook Alarm Security

Three-party Service (TSA) and Alarm Security Allowed (ASCA) Classes of Service are mutually exclusive. A set assigned TSA Class of Service cannot also be assigned ASCA Class of Service, and vice versa; a set assigned ASCA Class of Service cannot also be assigned TSA Class of Service.

Override, Enhanced

With Priority Override (POVR) equipped, there is a slight change in Multi-Party Operations functionality. When a consultation call is made without POVR equipped, and the telephone being called is busy, a recall returns to the party on hold without dialing a control digit. However, if POVR is equipped, a control digit must be dialed. Any control digit releases the busy call and returns to the call on hold.

Paging

Users of analog (500/2500 type) telephones cannot make a consultation call while connected to a paging trunk.

Recall to Same Attendant

Users of analog (500/2500 type) telephones can perform an attendant recall during a two-party connection by performing a switchhook flash and then dialing the attendant DN.

Recorded Telephone Dictation

Users of analog (500/2500 type) telephones cannot make a consultation call while connected to a dictation trunk.

Ring Again

When a TSA Class of Service analog (500/2500 type) telephone with a call on hold encounters Busy Tone, Ring Again is not possible.

Slow Answer Recall Enhancement

The Call Waiting Recall and Camp-on Waiting Recall enhancements take precedence over Attendant Recall Splitting (ATS), Secrecy (SYA), Enhanced Secrecy (EHS), and Multiple Party Operations.

Slow Answer Recall for Transferred External Trunks

The Multiple Party Operation recall can only be applied in a standalone environment, and therefore does not interact with this feature.

Stored Number Redial

For analog (500/2500 type) telephones with TSA Class of Service, the current LNR number can be stored only after the Consultation connection is completely released. Save Number Redial (SNR) is possible whenever Dial Tone or Special Dial Tone is given.

Tone to Last Party

When the MPO package is equipped, Tone to Last Party is not provided.

Trunk to Trunk Connection

In a standalone environment, the RGNA prompt in the Customer Data Block will be used when an external trunk is transferred on ringing and the called party does not answer. In a network environment, the RTIM timer value in the Customer Data Block will be used for slow answer recall.

Feature packaging

The basic Multi-Party Operations features are packaged under Multi-Party Operations (MPO) package 141.

For enhanced functionality of the Multi-Party Operations feature, the Flexible Tones and Cadences (FTC) package 125 is required.

Feature implementation

LD 10 – This overlay is modified to allow analog (500/2500 type) telephones to be assigned Three party Service Allowed (TSA) Class of Service.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	500	Type of telephone.
...		
CLS	TSA	Three-party Class of Service Allowed. TSA and ASCA are mutually exclusive (i.e., if TSA is assigned then ASCA will not be allowed, and vice versa). TSA interacts with XFA in the following manner. If the set has XFA (Call Transfer Allowed) Class of Service and the administrator then assigns TSA (Three-party Service Allowed), XFA (Call Transfer Allowed) is automatically set to XFD (Call Transfer Denied) and Three-party Service is then allowed. Conversely if the set has TSA Class of Service assigned and the administrator then assigns XFA, Three-party Service is removed and Call Transfer is allowed. The last Class of Service entered overwrites the previously entered Class of Service of the same category (i.e., if both XFA and TSA are entered in that order, TSA is the Class of Service that is accepted.)
...		

LD 15 – This overlay is modified to incorporate the following options if the MPO package is equipped:

— mandatory recall

— Control Digit Time Out

— treatment of switchhook flash

— programmable Control Digits

— alternative Consultation connection disconnect treatment, and

— mis-operation options.

To implement these options, the following prompts are added to LD 15:

Prompt	Response	Description
REQ	CHG, NEW	Change, or add.
TYPE	CDB MPO	Customer Data Block. Release 21 gate opener.
...		
- FMOP	(NO) YES	Flexible Mis-Operation Parameters. NO – (default) Do not change Flexible Mis-Operation Parameters. YES – Change Flexible Mis-Operation Parameters.

- - RGNA	xxxyyy	Ring No Answer. Enter treatment for Call Transfer Ring No Answer cases, where: xxx is the treatment for internal parties, and yyy is the treatment for external parties.
	(STD)(STD)	– (default for internal and external parties) Standard treatment
	AARAAR	– Attendant After Recall: Recall transferring (controller) set for RCY1 number of ring cycles. If transferring set does not answer within RCY1 ring cycles route call to an attendant.
	ATNATN	– Attendant: Route call to an attendant.
	DARDAR	– Disconnect After Recall: Recall transferring (controller) set for RCY1 number of ring cycles. If transferring set does not answer within RCY1 ring cycles disconnect call.
	DISDIS	– Disconnect: disconnect call.
	OVFOVF	– Overflow tone: Call is given Overflow Tone.
- - AOCS	xxxyyy	All Other Cases. Enter treatment for Call Transfer cases other than Ring No Answer: xxx is the treatment for internal parties and yyy is the treatment for external parties.
	AARAAR	– Attendant After Recall: Recall transferring (controller) set for RCY1 number of ring cycles. If transferring set does not answer within RCY1 ring cycles route call to an attendant.
	ATN(ATN)	– (default treatment for external parties) Attendant: Route call to an attendant.
	DARDAR	– Disconnect After Recall: Recall transferring (controller) set for RCY1 number of ring cycles. If transferring set does not answer within RCY1 ring cycles disconnect call.
	(DIS)DIS	– (default treatment for internal parties) Disconnect: disconnect call.
	OVFOVF	– Overflow tone: Call is given Overflow Tone.
	STDSTD	— Standard treatment ***Note:*** If entered in response to AOCS in LD 15, responses will be printed as DIS ATN in LD 21.

- - RCY1	1-(6)-15	Ring Cycles 1. Number of ring cycles (default is 6) a transferring (controlling) station is rung before routing to an attendant or disconnect occurs.
- - RCY2	1-(4)-15	Ring Cycles 2. Number of ring cycles (default is 4) target (transferred to) station is rung before Ring No Answer treatment is applied. Does not apply to AOCS.
- - RALL	(NO), YES	Recall NO – (default) Mandatory recall is not required prior to dialing Control Digits. YES – Mandatory recall is required prior to dialing Control Digits.
- - CDTO	2-(14)	Control Digit Time Out. Range is 2 - 14 seconds and inputs must be a multiple of 2, (i.e., 2, 4, 6, 8, 10, 12, or 14). 2 to 12 activates the optional time out treatment. 14 (default) activates the normal time out treatment.
- IFLS	(NO), YES	Ignore switchhook flash. NO – (default) Allows a switchhook flash, or dial "1", from an analog (500/2500 type) telephone to be interpreted as a Register Recall. YES – A switchhook flash, or dial "1", from an analog (500/2500 type) telephone will not be interpreted as a Register Recall. ***Note:*** If this option is selected, analog (500/2500 type) telephones should be equipped with a special Ground (Earth) Button.
- MHLD	(NO), YES	Manual Hold. NO – (default) Manual hold is not allowed. YES – Manual hold is allowed.

- PCDS	(NO), YES	Program Control Digits. YES – Allows user to alter default settings of Control Digits. NO – (default) Does not allow the alteration of the existing Control Digit settings. CCDO is the next prompt. Programming of control digits is not required. The default is NO. The defaults values for their respective functions are 1, 2 and 3. If YES then:
- - CNFD	0-(1)-9, *,#	Conference Digit. Prompted if response to PDCS was YES. Enter the Control Digit used to create, or add parties to, a conference. Default is 1.
- - TGLD	0-(2)-9, *,#	Toggle Digit. Prompted if response to PDCS was YES. Enter the Control Digit used to toggle, put active party on hold and connect to held party/parties. Default is 2.
- - DISD	0-(3)-9, *,#	Disconnect Digit. Prompted if response to PDCS was YES. Enter the Control Digit used to disconnect the active party and connect to the held party. Default digit is 3.
- CCDO	(NO), YES	Consultation Connection Disconnect Option. NO – (default) Alternative treatment is not applied to Consultation calls where one of the parties disconnects. YES – Alternative treatment is applied to Consultation calls where one of the parties disconnects.
- AFNO	(NO), YES	(Manual) Forced Camp-On Automatic.
- - ACNS		Attendant Clearing during Night Service. Prompted when the MPO package is equipped and MPOP and FMOP = YES.
	(NO) EXT ALL	No automatic treatment. External calls only. All calls.

LD 20 – This overlay is modified to print the Three-party Service (TSA) Class of Service if the MPO package is equipped.

LD 21 – This overlay is modified to print the Multi-Party Operations settings in the Customer Data Block if the MPO package is equipped.

LD 22 – This overlay is modified to print the Multi-Party Operations package (141) if it is equipped.

LD 56 – This overlay is used to specify all Tones and Cadences. LD 56 is modified to allow specification of Control Dial Tone and recall Tones and Cadences cadences for analog (500/2500 type) telephones and Meridian 1 proprietary telephones.

Prompt	**Response**	**Description**
REQ	CHG NEW PRT	Change, add, or print.
TYPE	FTC	Flexible Tones and Cadences data block.
TABLE	0-31	FTC table number.
...		
RING	YES	Change the ringing feature definitions.
...		
- PCAD	xxx	Recall of mis-operation ringing Cadence Enter Master Cadence (MCAD) table number that defines the ringing cadence for recall of mis-operation for analog (500/2500 type) telephones and Meridian Modular sets. Default is value assigned to NCAD.
- PBCS		Recall of mis-operation ringing tone and cadence for Meridian 1 proprietary telephones. Define recall of mis-operation tone and cadence for SL-1 and M1000 sets.

TDSH	i bb cc tt	Tone and Digit Switch Hexadecimal code. Prompted if Tone and Digit Switches (TDS) are configured in LD 17. Defaults are values assigned to NBCS.
XTON XCAD	0-255 0-255	Extended tone code. Extended cadence code. Respond to the XTON prompt with a value from 0 to 255, for the NT8D17 TDS Tone code. Respond to the XCAD prompt with a value from 0 to 255, for the NT8D17 TDS cadence code for FCAD. Prompted if system configured with Extended Conference and Tone and Digit Switches (XCT) in LD 17. Defaults are the values assigned to NBCS.
...		
HCCT	YES	Hardware Controlled Cadences and Tones.
...		
- CDT		Control Dial Tone. Define tone and cadence for Control Dial Tone.
TDSH	i bb cc tt	Tone and Digit Switch Hexadecimal code. Prompted if Tone and Digit Switches (TDS) configured in LD 17. Defaults are values assigned to DIAL.
XTON XCAD	0-255 0-255	Extended tone code. Extended cadence code. Respond to the XTON prompt with a value from 0 to 255, for the NT8D17 TDS Tone code. Respond to the XCAD prompt with a value from 0 to 255, for the NT8D17 TDS cadence code for FCAD. Prompted if system configured with Extended Conference and Tone and Digit Switches (XCT) in LD 17. Defaults are the values assigned to DIAL.

Note: Refer to *X11 data administration input/output guide including supplementary features* (Appendix 1 to 553-2311-311) for complete information regarding the administration of tones and cadences.

LD 81 – This overlay is modified to print the stations associated with Three-party Service Allowed (TSA) Class of Service if the MPO package is equipped.

LD 83 – This overlay is modified to include the TSA Class of Service, when sorting TN by Class of Service, if the MPO package is equipped.

Feature operation

Prior to describing the feature operation the following terms are defined to ensure there is no misunderstanding as to their meaning in terms of the Multi-Party Operations feature.

Active party – The party with which the controlling party has a Consultation connection.

Meridian 1 proprietary telephone – For the purpose of this document, this term is used to refer to standard SL-1 sets and to digital sets (M2000 series and M3000).

Bridged sets – The "Bridging" feature allows the same DN to appear on more than one single line telephone. Bridged sets share the same TN. Up to eight of these sets can be bridged, and a maximum of five of this group can be equipped with ringers. An incoming call rings all sets that have ringers connected and can be answered by any single line telephone user within the bridged group.

Controlling party – The "Controlling Party" is the party which optionally has the "Held Party" in the hold mode and the "Active Party" in the "Consultation Connection".

Consultation connection – When the controlling party and the active party are in conversation, they are said to be in "Consultation Connection".

Dial "1" – A pulse recognized as digit 1.

External party – Any CO, DID or TIE trunk (incoming or outgoing), connected to the system is considered an external party, regardless of the way the connection is established.

Flash Timer – The Flash Timer defines the flash period of a valid Switchhook Flash.

Held party – The Held party is the party put on hold (by the Controlling party).

Programmable Control Digit – A digit which is dialed by the controlling party, after the Consultation connection is established, to achieve certain functions of Three-party Service for an analog (500/2500 type) telephone.

Register Recall – A user request for service produced either by Switchhook Flash or by pressing the Ground Button or the Link button.

Switchhook Flash – An on/off-hook pulse which may be either a Register Recall signal or a Digit 1 depending on the conditions during which it occurs and on the flash timing.

Call Join

Call Join is available on any Meridian 1 proprietary telephone that is equipped with a Three-party (AO3) or Six-party (AO6) Conference key and at least one secondary DN or Call Waiting key.

The following describes the operation of Call Join:

1. If the user presses the AO3 or AO6 key during an active call with party A on DNx (DNx is any DN key, including Call Waiting), party A is placed on hold and Special Dial Tone is returned as normal. The user can dial another DN and conference as normal or the user can conference a held party B on DNy (DNy is any DN key, excluding DNx) by continuing as follows:

2. The user presses DNy during Special Dial Tone. This causes party B to be moved to the Conference key. DNy key is idled. The Conference key remains active and the user consults with party B.

3 When the user has finished consulting with party B, the user presses the Conference key a second time. Party A, party B and the user form a conference (subject to normal restrictions) on DNx. The Conference key is idled. If the user disconnects during the conference, party A is transferred to party B, subject to normal restrictions.

The conference can be enlarged by operating the AO6 key either as described above to add a held party to the conference, or as normal to conference a dialed party.

Note 1: The DNx or secondary DN key can be any Meridian 1 proprietary telephone key capable of holding an independent Directory Number.

Note 2: If the Call Waiting is a Group Call, that call cannot be joined.

Note 3: M2317 and M3000 set soft keys may not display Procedure 1 correctly.

Analog (500/2500 type) telephone features

Multi-Party Operations introduces Three-party Service Allowed (TSA) Class of Service. Analog (500/2500 type) telephones can now be assigned TSA Class of Service and either C6D (Conference 6-party Denied) or C6A (Conference 6-party Allowed) Class of Service. Analog (500/2500 type) telephone operation is not changed for XFD or XFA Classes of Service.

Three-party Service permits the user to toggle, release or form a three-party conference through the use of Programmable Control Digits.

The combination of TSA and Conference 6-party (C6A) Classes of Service extend the operation of Three-party Service so as to permit the user to enlarge the three-party conference by consulting and selectively adding members through the use of Programmable Control Digits.

The following sections describe Three-party Service (TSA Class of Service).

Establishing a Consultation connection

If the user requests a Register Recall during any established two-party connection, excluding calls to Dictation or Paging trunks or to an attendant, the call is placed on hold and Special Dial Tone is returned. The user can dial a second party for Consultation.

If the controlling party goes on-hook before the second call is established (that is, when the transferred station is ringing, the call is treated as per Mis-operation of Call Transfer).

When the second call is established, the user becomes the controlling party of the "Consultation" connection. The user can modify the connection through the use of a Programmable Control Digit.

Dialing a Control Digit from a Dial Impulse analog (500/2500 type) telephone with DIP or DTN Class of Service

After the consultation connection is established, the controlling party can dial a Programmable Control Digit. Here, if RALL = NO, both sets with DIP and DTN Class of Service dialing using dial impulses can dial the programmable control digits without performing the recall. However, for a dial impulse sets with DTN Class of Service the mode of dialing control digits depends upon how the set has setup the consultation call. If the set has used pulse dialing, then the control digits are recognized without recall. If the set has used touchtone dialing, Register Recall is mandatory.

If RALL = YES, a register recall must be performed prior to dialing a control digit, regardless of the set's Class of Service.

1 Dialing the Conference (CNFD) Control Digit produces a three-party conference between the user, held and active parties. During the Conference connection, all parties are restricted from using Call Transfer, Three-party Service and Three-party Conference features (unless the user has C6A Class of Service). If the user goes on-hook during the conference, the remaining parties stay connected as a normal two-party call, subject to normal restrictions.

2 Dialing the Toggle Control Digit (TGLD) exchanges active and held parties. During the Consultation connection, the controlling party is restricted from adding other parties to the call, and the non-controlling parties are restricted from using the Call Transfer, Conference and Three-party Service features.

3 Dialing the Disconnect Active Control Digit (DISD) releases the active party. The connection to the held party is automatically restored as a normal two-party connection. Either party can initiate another Consultation or Conference connection, subject to normal restrictions.

If the user dials any other digit, the connection to the active party is restored and the held party remains on hold.

Dialing a Control Digit from a Dual-tone Multifrequency analog (500/2500 type) telephone with DTN Class of Service

After the Consultation connection is established, the controlling party can dial a Control Digit. If the controlling party is a Dual-tone Multifrequency (DTMF) analog (500/2500 type) telephone with DTN Class of Service, a Register Recall must precede the Programmable Control Digit.

When the controlling party performs a Register Recall, the speechpath to the active party is removed. If no Digitone Receivers (DTRs) are available, no tone is given and the active party is reconnected. If a DTR is found, a new tone, Control Dial Tone, is given to the controlling party. The cadence, level and frequency of Control Dial Tone are flexible and defined on a per-customer basis.

During Control Dial Tone, the user can dial a Programmable Control Digit.

If a disconnect signal is received from the held party during Control Dial Tone, or if the user does not dial a Programmable Control Digit within 15 seconds, the DTR is removed and Overflow Tone is given for 14 seconds. During this time, the controlling party can restore the connection to the active party by performing a switchhook flash. At the end of Overflow Tone, the active party is reconnected and the held party (if still connected) remains on hold.

If the user performs a switchhook flash during Control Dial Tone, the connection with the active party is restored and the held party remains on hold.

Dialing a Control Digit from a Bridged Set

If Dial Impulse analog (500/2500 type) telephones and DTMF analog (500/2500 type) telephones are bridged and assigned DTN Class of Service, the operation depends on whether the Consultation connection was set up using Dial Impulse or DTMF.

If the Consultation connection was set up using Dial Impulse, only Dial Impulse analog (500/2500 type) telephone users can dial a Programmable Control Digit. If the Consultation connection was set up using DTMF, only DTMF analog (500/2500 type) telephone users can dial a Programmable Control Digit.

Any dial pulses or Register Recalls are recognized only if all other sets on the bridged line are on-hook. A Register Recall performed by using a Ground Button is also recognized.

Controlling party actions

The following table summarizes the affect on Consultation connections when controlling parties with XFA or TSA Class of Service perform the following actions:

Table 109
Control Digit Functions

Controlling party action	System Response Class of Service	
	XFA	TSA
Dial CNFD	Conference	Conference
Dial TGLD	Conference	Toggle
Dial DISD	Conference	Release Active Party
On-hook	Transfer	Disconnect

Note 1: Dial Impulse analog (500/2500 type) telephones with DIP Class of Service are required to issue a Register Recall prior to dialing Control Digits if RALL = YES.

Note 2: If Control Dial Time Out is any value other than the default (14), then the time out results in the same action as if DISD had been dialed.

Note 3: If CCDO is YES, Call Transfer takes place when the controlling party goes on-hook during a consultation connection. This is similar to XFA operation.

Consultation Call Disconnect

Active Party Disconnects

If the disconnect during Consultation connection option chosen is the default (CCDO = NO), after a Consultation connection has been established, and the active party disconnects if:

1. The active party is internal to the PBX or is external and a disconnect signal is received by the PBX, the held party is reconnected for a normal two-party connection.
2. The active party is external to the PBX and a disconnect signal is not received by the PBX, then the controlling party is able to release the disconnected trunk by dialing the Disconnect Active (DISD) Programmable Control Digit. The connection to the remaining party then becomes a normal two-party connection.

If the disconnect during Consultation connection option chosen is to give the alternative treatment (CCDO = YES), then if the active party goes on-hook during an enquiry call, the controlling party is given Overflow Tone. On tone time out or Register Recall, the held party is reconnected. If the controlling party goes on-hook during Overflow Tone, the call is treated as in Controlling Party Disconnects.

Held Party Disconnects

If the disconnect during Consultation connection option chosen is the default, after a Consultation connection has been established, the held party disconnects if:

1. The held party is internal to the PBX or is external and a disconnect signal is received by the PBX, then the connection with the active party becomes a normal two-party connection.
2. The held party is external to the PBX and a disconnect signal is not received by the PBX, due to the fact that the remaining connection is effectively a two-party connection, the trunk to which the departed party was connected is still on hold. The controlling party can release the disconnected trunk by dialing the Toggle (TGLD) Programmable Control Digit (to hold the active party and activate the connection to the disconnect trunk) and then dialing the Disconnect Active (DISD) Programmable Control Digit (to release the trunk). The connection to the remaining party becomes a normal two-party connection. If the controlling party goes on-hook with the disconnected trunk on hold, the set is rung back.

In the case of Procedure 1, when a Dial Impulse analog (500/2500 type) telephone with DIP Class of Service user dials a Programmable Control Digit during the active call, Special Dial Tone is returned, indicating that the held party has disconnected. Similarly, when a DTMF analog (500/2500 type) telephone with DTN Class of Service user performs a switchhook flash during the active call (expecting to receive Control Dial Tone), Special Dial Tone is returned, indicating that the held party has disconnected.

During Special Dial Tone, the controlling party has the option of dialing a DN to set up another Consultation connection, or of resuming the normal two-party connection. The latter is achieved by performing a Register Recall with a duration greater than 150 milliseconds and less than the maximum flash time (a short Register Recall would be mistaken for a digit "1"). A dial "1" from a Dial Impulse analog (500/2500 type) telephone with DIP Class of Service cannot be used to simulate the flash, as the digit "1" may be the first digit of a DN. The user can do a valid switchhook flash during the middle of dialing a DN and be returned back to he held party. The only restriction is that the switchhook flash must be unambiguous (that is, the duration of the switchhook flash is greater than the digit "1" duration).

If a 2500 set recalls during a consultation connection and the held party has disconnected with the held party being an internal party or the Meridian 1 has received a disconnect signal, special dial tone is returned instead of control dial tone. This is similar to CCDO = NO.

If RALL = YES, the above operation also applies to 500 sets.

With RALL = NO and a 500 set (dial impulse) dials a control digit other than DISD, the set is given overflow tone indicating that the held party has disconnected. If the 500 set dials the DISD control digit, the active party is disconnected and the control party gets overflow tone.

Controlling Party Disconnects

If the disconnect during Consultation connection option chosen is default, then if the controlling party goes on-hook during the Consultation connection, it is considered as a mis-operation of All Other Cases type (AOCS) and the active party is released.

DIS, ATN, AAR, DAR, and OVF options are available for both internal and external parties. If the held party is internal to the PBX, the held party is optionally (DIS) released also. If the held party is external, the controlling set is optionally (AAR) rung back immediately. The external party does not receive Ringback Tone while the controlling set is being rung.

If the controlling party answers, the external party is connected for a normal two-party connection. If the controlling party does not answer within the optional ring cycles (RCY1) for any call (regardless of whether the set has FND or FNA Class of Service), the controlling station is idled while the external party receives Ringback Tone and is optionally routed to the attendant and appears on the CFNA Incoming Call Indicator. Other options are also available.

If the disconnect during Consultation connection option chosen is to give the alternative treatment, then if the controlling party goes on-hook during conversation with the active party, the call is transferred (as current operation with XFA Class of Service on the station).

Six-party Conference

The combination of C6A and TSA Classes of Service, provides an enhancement to the Six-party Conference feature where the user can perform a Register Recall during the Conference connection, dial a consulted party and then dial a Programmable Control Digit to toggle, release, or add the consulted party to the conference. The following describes the sequence of events required of a analog (500/2500 type) telephone with C6A and TSA Classes of Service to set up a multi-party conference:

1 During a normal two-party connection with party A, the user performs a Register Recall and dials party B. The user becomes the controlling party of the Consultation connection.

2 The user dials the Conference (CNFD) Programmable Control Digit to form a three-party conference. The Consultation connection becomes a Conference connection.

3 The user performs a Register Recall during the Conference connection. The conference is placed on hold (the other parties in the conference remain connected) and Special Dial Tone is returned. The normal timing and mis-operation procedures apply while setting up the Consultation call. The user dials party C. When the Consultation call is established, the user becomes the controlling party of the new Consultation connection.

4 The user can dial a Programmable Control Digit which is interpreted as follows:

— Dialing CNFD causes the consulted party to be added to the conference, as shown in Procedure 2. The Consultation connection becomes a Conference connection.

— Dialing TGLD causes the consulted party to be placed on hold and the conference to be reconnected. The user can toggle between the conference and the consulted party in this manner.

— Dialing DISD causes the consulted party to be disconnected. The Conference connection is restored.

The user can repeat Steps 3 and 4 to add parties to the conference. If the user goes on-hook during the Consultation connection, the consulted party is released and the conference stays connected, subject to normal restrictions. Six-party Conference Enhancement for analog (500/2500 type) telephones follows the same operation as the existing Six-party Conference feature with respect to mis-operation, access and connection restrictions.

Recovery of mis-operation during Call Transfer

Call Transfer with Ring No Answer (RGNA)

RGNA is applicable only when the user transfers a call while the active party is still in ringing state. All other types of mis-operation are handled as AOCS mis-operations.

Call treatment is then determined by the response to the RGNA prompt in LD 15. Following is the list of the responses to the RGNA prompt and their resulting treatment:

1. STD (Standard) – The operation as it was prior to the introduction of the MPO feature.
2. ATN (Attendant) – The transferred party is routed to the attendant if the target (transferred to) station, after having rung for an optional number of ring cycles (RCY2), has not answered the call. The call is rerouted to the attendant as a Call Forward No Answer (CFNA) and is presented on the FNA Incoming Call Indicator (ICI), the call is then treated as a regular CFNA call to the attendant.
3. If the transferred call was a Consultation connection the transferred party is disconnected and the held party is routed to an attendant and presented as a Recall on the RLL ICI.
4. DAR (Disconnect After Recall) – The target station rings for an optional number of ring cycles (RCY2). If the call is not answered during this time, the transferred party recalls the transferring (controlling) station. The transferring station rings for an optional number of ring cycles (RCY1), with recall ringing cadence. If the transferring station does not answer during this time, the transferred party is disconnected.

5 If the transferred call was a Consultation connection then the held party is retrieved and treated as defined by its type (internal or external) and the treatment selected. If the treatment selected is ATN or AAR the held party is routed to an attendant and presented as a Recall on the RLL ICI. If the treatment selected is DAR or DIS, the party is disconnected.

6 If the transferring station became busy before recall, the transferred party is disconnected immediately.

7 AAR (Attendant After Recall) – This option is similar to the DAR option, except that after the optional number of ringing cycles (RCY1) the transferred party is routed to an attendant as a Call Forward No Answer (CFNA) recall and is presented on the CFN ICI.

8 If the transferred call was a Consultation connection then the held party is retrieved and treated as defined by its type (internal or external) and the treatment selected. If the treatment selected is ATN or AAR the held party is routed to an attendant and presented as a Recall on the RLL ICI. If the treatment selected is DAR or DIS, the party is disconnected.

9 If the transferring station became busy before recall, the transferred party is routed to attendant immediately.

10 OVF (Overflow) – Overflow Tone is given to the transferred party after the optional number of ring cycles (RCY2).

11 If the transferred call was a Consultation connection, the transferred party is disconnected and the held party is given Overflow Tone.

12 DIS (Disconnect) – The transferred party is disconnected after the optional number of ring cycles (RCY2).

13 If the transferred call was a Consultation connection the transferred party and held party are disconnected.

Note: The ring cycles are counted from the time the transfer has been completed (analog (500/2500 type) telephone has gone on-hook or Meridian 1 proprietary telephone has pressed the TRN key for the second time).

This feature applies to both external and internal calls, transferred by station users to another station. The feature does not apply to calls transferred to the attendant, or extended by the attendant.

Mis-operation during Call Transfer – All Other Cases (AOCS)

This section describes mis-operation during Call Transfer for All Other Cases (AOCS) and their default options. Similar options as for Ring No Answer (RGNA) are available for AOCS. The only difference being that the ringing cycle (RCY2) is not valid for AOCS.

Call Transfer to a Busy Station

If an analog (500/2500 type) telephone user tries to transfer a call to a busy station, Busy Tone is returned during the Consultation connection. If the user then goes on-hook to complete the Transfer operation and if the held party is an external trunk, the external trunk is routed automatically to the attendant as an Intercept Recall. The call is then treated as a regular Intercept Recall call to the attendant.

If the held party is an internal call, it is disconnected.

Call Transfer to Intercept Treatment

While using the Call Transfer feature, the analog (500/2500 type) telephone user may be intercepted while dialing the third party due to any of the following illegal dialing situations:

1 Dialing a vacant number.

2 Dialing a number of a terminal in the maintenance busy or RPE failure state.

3 Access denied.

4 Code Restriction or Toll Restriction.

5 Invalid, restricted, or blocked Network Automatic Route Selection (NARS) or Basic Automatic Route Selection (BARS) calls.

In any of the above cases, while involved in the Consultation connection (according to the selected customer option) the user is:

— given Overflow Tone

— given an intercept recorded announcement or

— routed to the attendant

If the user goes on-hook while connected to Overflow Tone or recorded announcement, and if the held party is an external trunk, the external trunk is routed to the attendant as an Intercept Recall. The call is then treated as a regular Intercept Recall to the attendant.

If the MPO package is equipped and the user waits until time out occurs while connected to Overflow Tone or a recorded announcement, the held party is reconnected to the station user, and the call is treated as a regular two-party call again.

If the MPO package is not equipped, and the user waits until time out occurs while connected to Overflow Tone or a recorded announcement, both the internal and external calls are disconnected.

Unsuccessful Transfer Connection

While transferring an external trunk to another destination from an analog (500/2500 type) telephone, if network blocking prevents the completion of the Call Transfer or if the controlling party dials the access code of a busy trunk route, the controlling party receives Overflow Tone during the Consultation connection. If the analog (500/2500 type) telephone user goes on-hook in spite of the blocking indication, the external trunk is routed to the attendant as an Intercept Recall. At this point, the call is treated as a regular Intercept Recall to the attendant.

Call Transfer on Partial Dialing

If an analog (500/2500 type) telephone user dials an incomplete number as a third party and attempts to complete the Transfer operation by going on-hook, and if the held party is an external trunk, the external trunk is routed to the attendant as an Intercept Recall. The call is then treated as a regular Intercept Recall to the attendant.

Disconnect Situations during Consultation

If the analog (500/2500 type) telephone user (the controlling party) disconnects while in the Consultation state, the call is transferred as normal. However, if the new connection is not possible (for example, due to trunk-to-trunk connection restrictions), and if the held party is external, then this external party is routed to the attendant as an Intercept Recall. The call is then treated as a regular Intercept Recall to the attendant.

Also, if the analog (500/2500 type) telephone user (the controlling party) disconnects while connected to Dial Tone, and if the held party is external, then this external party is routed to the attendant as an Intercept Recall. The call is then treated as a regular Intercept Recall to the attendant.

If one of the other parties in the call disconnects, the following occurs:

— If the held party disconnects, the controlling party receives no indication until the hook switch is flashed to establish a conference. At that time Dial Tone is returned instead of all three parties creating a conference. The call is treated as a normal two-party call from the time the held party disconnects.

— While an external party is in the Consultation hold state, if the party being consulted disconnects followed by the controlling party disconnect, then the held party is routed automatically to the attendant as an Intercept Recall. The call is then treated as a regular Intercept Recall to the attendant.

Mis-operation during Control Dial Tone

The treatment given depends upon the type of active party. If the active party is internal, the internal option is also applied to the held party (for example, if for internal calls AOCS is DIS ATTN, the held call even though external will also be disconnected). The mis-operation option selected in this case is solely dependent upon the type of active call (internal or external), and the related mis-operation option. This option is consistently applied to the held, as well as the active party.

With the Consultation Connection Disconnect Option (CCDO) in LD 15 not selected, if an analog (500/2500 type) telephone user (the controlling party) disconnects while receiving Control Dial Tone in the Consultation state, internal held parties are disconnected while external parties are routed to the attendant as Intercept Recalls. The external calls are then treated as a regular Intercept Recalls to the attendant.

With CCDO selected, if an analog (500/2500 type) telephone user (the controlling party) disconnects while receiving Control Dial Tone in the Consultation state the held parties are given treatment as defined by the responses to the All Other Cases (AOCS) prompt in LD 15.

Mis-operation Treatment Options

A number of mis-operation treatment options are made available both for internal and external calls. These treatment options are available for Ring No Answer (RGNA) and for All Other Cases (AOCS). The following are the cases for AOCS:

- — Call Transfer to Intercept Treatment for:
  - Call Transfer to busy station
  - Dialing a vacant number
  - Terminal is in maintenance busy
  - RPE failure state
  - Access denial
  - Code or Toll restricted set
  - Network blocking
  - Invalid, restricted and blocked Network Automatic Route Selection (NARS)/Basic Automatic Route Selection (BARS) calls
  - Partial dialing
  - Trunk-to-trunk connection restrictions
  - Inter-tenant blocking
  - During reception of announcements, and
  - During reception of tones (Control, Special),
- — Call Transfer while Dial Tone is being heard
- — Call Transfer before completing dialing
- — Call Transfer during outpulsing of digits on a trunk, and
- — Controlling party goes on-hook during Consultation connection (CCDO = NO).

Recall of mis-operation Ringing Cadence Option

When a transferring set is rung back after Call Transfer mis-operation, then Recall of mis-operation ringing cadence is optionally given to this set. Two optional Recall of mis-operation cadences, one for analog (500/2500 type) telephone and Meridian Modular sets (PCAD) and another for SL-1 and M1000 series sets (PBCS), are optionally selectable (in LD 56). The default Recall of mis-operation ringing cadence is the current Ringing Tone or cadence.

Introduced in X11 Release:	16.67G
Networking:	No

Multi-Party Operations Enhancements

The following enhancements pertain to the Three-party Service capability of Multi-Party Operations. Refer to the Multi-Party Operations feature description contained in this document for a description of Three-party Service.

Patience Tone

The controlling party may modify a Consultation connection by performing a Register Recall and then entering a Control Digit. During the call modification, this enhancement provides a "Patience" tone to the party on Consultation hold, rather than silence.

Ringback to external parties after misoperation

If the controlling party goes on-hook as a misoperation, the controlling set is rerung immediately. This enhancement allows the external party to receive ringback tone while the controlling party is rerung after mis-operation.

Operating parameters

There are no feature requirements.

Feature interactions

Attendant Break-in

Attendant Break-in is not allowed to a connection in which a party is receiving Patience Tone or recall of mis-operation ringback.

Call Transfer

A party receiving Patience Tone or recall of mis-operation ringback is not able to Call Transfer.

Call Waiting

An analog (500/2500 type) telephone cannot have Call Waiting during Patience tone.

Camp-on
Periodic Camp-on

While Camp-on and Periodic Camp-on are allowed on a party receiving Patience Tone, Camp-on tone and Periodic Camp-on tone are not applied to the party during Patience tone. However, Camp-on tone and Periodic Camp-on tone are applied when the speechpath has been reestablished.

Conference

Patience tone or recall of mis-operation ringback are not applied to a conference party.

End-to-end Signaling

A party receiving Patience Tone or recall of mis-operation ringback is not able to invoke End-to-end Signaling.

Multi-Party Operations

Usually the party on Consultation hold receives silence, with this improvement it will receive Patience tone.

After a mis-operation when the controlling party is rerung the far end receives silence, with this improvement it will receive ringback tone.

Feature packaging

These enhancements are packaged as part of the Supplementary Features (SUPP) package 131.

French Type Approval (FRTA) package 197 is also required to provide ringback tone to the held party while the controlling party is being rerung.

Feature implementation

LD 56 – This overlay is used to specify all tones and cadences. LD 56 is modified to allow specification of Patience tone:

Prompt	Response	Description
REQ	CHG, NEW, PRT	...
TYPE	FTC	...
...		
HCCT	YES	
_TLPT	...	
_PATI		Patience tone. Define Patience tone and cadence.
TDSH	i bb cc tt	Tone and Digit Switch Hexadecimal code. Prompted if Tone and Digit Switch (TDS) is configured in LD 17. Default is (0000) no tone.
XTON XCAD	(0)-255 (0)-255	Extended Tone code. Extended Cadence code. Respond to the XTON prompt with a value from 0 to 255, for the NT8D17 TDS tone code. Default is 0. Respond to the XCAD prompt with a value from 0 to 255, for the NT8D17 TDS cadence code for FCAD. Default is 0. Prompted if system configured with Extended Conference and Tone and Digit Switches (XCT) in LD 17. Default is no tone.
...		

Note: Refer to *X11 input/output guide* for complete information regarding the administration of tones and cadences.

Feature operation

Patience Tone to Consultation Held party during Control Dial Tone

To initiate Three-party Service analog (500/2500 type) telephones must perform a Register Recall, (i.e. Switchhook Flash).

When the controlling party has established a Consultation connection, there is a call on hold and the Consultation connection is active. The controlling party can modify the connection through the use of a Control Digit.

To modify the call the controlling party performs a Register Recall, if the response to RALL in LD 15 is YES, to receive Control Dial Tone for 15 seconds. If no digit is dialed within 15 seconds the controlling set then receives Overflow Tone. If no digit is dialed, the controlling set is eventually put in lockout state.

Current operation is when a controlling party performs the Register Recall, the speechpath to the consulted party is removed, and the consulted party receives silence.

This enhancement allows a Patience Tone to be given to the consulted party instead on silence while the speechpath is removed.

Ringback sent when the controlling party is rerung after a mis-operation

Current operation is when a controlling party goes on-hook and the on-hook constitutes a mis-operation, the initial held call or the held consultation party may re-ring the controlling set immediately if the appropriate option (either AAR or DAR) is active. The external party does not receive ringback tone while the controlling set is being rung.

This enhancement allows a ringback tone to be provided to the external party when the controlling set is being rerung.

Introduced in X11 Release:	All
Networking:	No

Multiple Appearance Directory Number Redirection Prime

With X11 Release 18 and later, Multiple Appearance Directory Number (DN) Redirection Prime (MARP) standardizes call redirection on Multiple Appearance DNs (MADNs) by using a service changeable Multiple Appearance DN Redirection Prime Terminal Number (MARP TN).

Each defined Single or Multiple Appearance DN has only one associated MARP TN. When a call redirection feature activated against a DN needs Terminal Number (TN) specific information, the MARP TN is used to determine feature operation. Call redirection always refers to the MARP TN.

MARP provides consistent operation for the following call redirection features:

— Call Forward All Calls

— Call Forward Busy

— Call Forward No Answer, and

— Hunting.

Operating parameters

Short Hunt takes precedence over MARP TN directions.

MARP is activated in LD 17. If MARP is not active, call redirection occurs according to the pre-X11 Release 18 algorithms. All the MARP prompts and messages appear even if MARP is not active. MARP TNs can still be added, assigned, and changed. Refer to specific call redirection modules in this document for details regarding the pre-X11 Release 18 algorithms.

The MARP TN is defined in LD 10 or LD 11. When activated, only the MARP TN is used to determine call redirection.

If MARP is not activated, the overlays listed have this message printed: "MARP NOT ACTIVATED." The message appears only once, when the overlay is loaded. When MARP is active, no message appears. The overlays affected are: LDs 10, 11, 20, 22, 25, 80, 81, 82, and 83.

When MARP is activated in Service Change (MARP = YES), calls are immediately directed according to the MARP TN. There is no need to SYSLOAD.

Every Single or Multiple Appearance DN has a MARP TN. MARP TNs are also defined for Data DNs, optional incoming two-way Hot Line DNs, and ringing and nonringing Private Line DNs. Automatic Call Distribution (ACD) DNs are not assigned MARP TNs.

New systems are installed with MARP activated. MARP TNs are assigned to all Single and Multiple Appearance DNs. Call redirection follows the MARP TN assignments.

Conversion

When converting pre-X11 Release 18 software to X11 Release 18 or later, a MARP TN is automatically assigned for each Single and Multiple Appearance DN. This conversion does not activate MARP. Call redirection operates according to the pre-X11 Release 18 algorithms. All the MARP prompts and messages appear even if MARP is not active. MARP TNs can still be added, assigned, and changed.

When operating on X11 Release 18, and converting to an up-issue, the MARP TN assignments remain. If MARP was activated, it retains that activity following the up-issue. If MARP was deactivated, that status is also maintained following the up-issue.

MARP TNs assigned at Service Change

Each DN must have an associated MARP TN. After a Service Change or a telephone relocation, the system assigns a MARP TN to the DN in the following situations:

— The MARP TN containing the DN is removed.

— The DN appearance on its MARP TN is changed to another DN.

— The DN appearance on its MARP TN is no longer the redirection prime.

The "TN list" refers to the list of TNs that appears when you print the DN block in LD 20 or LD 22 (TYPE = DNB). To determine the order in which your TNs appear, print out the DN block.

When assigning MARP TNs during Service Change, the system conducts a search beginning at the top of the TN list for the first appearance of the DN as the Prime DN. The MARP TN is assigned based on the following:

1 The first TN found with a primary appearance of the DN is assigned as the MARP TN.

2 If no primary appearance of the DN is found, the first TN encountered with a secondary appearance of the DN is assigned as the MARP TN.

MARP TNs assigned at conversion and SYSLOAD

When converting to X11 Release 18, a MARP TN is automatically assigned to each DN at SYSLOAD. The MARP TNs are assigned to the DNs based on the following:

1 The lowest numerical TN with a primary appearance of the DN is assigned as the MARP TN.

2 If no primary appearance of the DN is found, the lowest numerical TN with a secondary appearance of the DN is assigned as the MARP TN.

CAUTION

MARP assignments made during conversion may change the manner in which calls are redirected. Refer to the individual call redirection modules in this document for details of the pre-X11 Release 18 algorithms.

Feature interactions

Attendant Administration

MARP TNs cannot be added, moved, or deleted with Attendant Administration. The DN information that displays on the console includes the MARP designation if applicable.

Attendant administration activities, like changing key assignments or DN appearance, may change MARP TN assignments. If so, CSC102 appears on the teletype (TTY) indicating a new default MARP TN, as follows:

CSC102 DN nnnn NEW MARP l s c u

where

nnnn = the DN associated with the MARP TN
l s c u = the new MARP TN assigned to DN nnnn

Attendant and Network-Wide Remote Call Forward (RCFW)

The RCFW feature operation applies only to one prime DN of a Multiple Appearance DN. If multiple stations are configured with the same prime DN, the set-based network RCFW feature operation is the same as the standalone RCFW feature operation.

If multiple stations are assigned the same prime DN and station control password (SCPW), the RCFW operation applies to the station to which the MARP TN is assigned. If none of the stations is configured as the MARP TN for that prime DN, the Remote Call Forward Activate and Deactivate Flexible Feature Codes (FFCs) apply to all stations matching the DN and SCPW. Remote Call Forward Verify applies to the station according to MADN call presentation priority, placing the station with the last service change at the end of the list.

The attendant-based RCFW operation applies to the station with the MARP TN of the DN entered.

Attendant Break-In

The attendant may get a busy tone if all the telephones with the required DN are busy. Attendant Break-In permits the attendant to break in to the connection with the least restricted TN. Where more than one TN exists that meets this criterion, Break-In chooses the one at the bottom of the DN block.

Automatic Set Relocation
Modular Telephone Relocation

When Automatic Set Relocation is used to move a telephone, the telephone's MARP designations are maintained. During the relocation, a temporary MARP TN is assigned. The original MARP TN is restored when the telephone relocates.

When a telephone leaves the system due to set relocation, the following Customer Service Change (CSC) message appears:

CSC010 x y

where
x = old TN (l s c u) for the telephone
y = ID code entered

The following Service Change (SCH) message appears for any MARP TN reassignment:

SCH5524 DN nnnn NEW MARP l s c u

where
nnnn = the DN associated with the MARP TN
l s c u = the new default MARP for DN nnnn

The History File can be configured to store these messages until a printout is requested.

When a telephone reenters the system, the following message appears:

CSC011 x y

where
x = old TN (l s c u) for the telephone
y = new TN (l s c u) for the telephone

The following message appears again for each changed TN:

SCH5524 DN nnnn NEW MARP l s c u

where
nnnn = the DN associated with the MARP TN
l s c u = the new MARP TN assigned to DN nnnn

Automatic Call Distribution

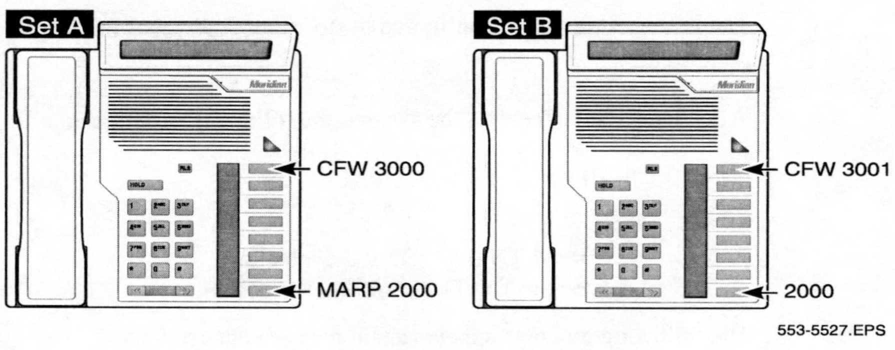
Automatic Call Distribution (ACD) DNs are not assigned MARP TNs. Agent Individual DNs (IDNs) are assigned MARP TNs.

Call Forward All Calls

If CFW is active for a DN, incoming calls are forwarded if a TN is found that has CFW enabled and is a single appearance or a prime multiple appearance of that DN (according to existing operation). The MARP TN is always checked first to meet these criteria. When the requirements are met, the system uses the information associated with the MARP TN to redirect the call.

If the MARP TN is not a prime appearance but does have CFW enabled, a search is made for a telephone with a prime appearance of that DN with CFW enabled. When a TN is found, the call is redirected according to the MARP TN's parameters. If the MARP TN is not a prime appearance and does not have CFW enabled, the system searches for a prime appearance with CFW enabled. The incoming call is forwarded according to the other telephone's instructions (not the MARP TN's), as shown in Figure 62.

Figure 62
CFW and MARP



CFW DN on Telephone A is DN 3000. CFW DN on Telephone B is DN 3001.

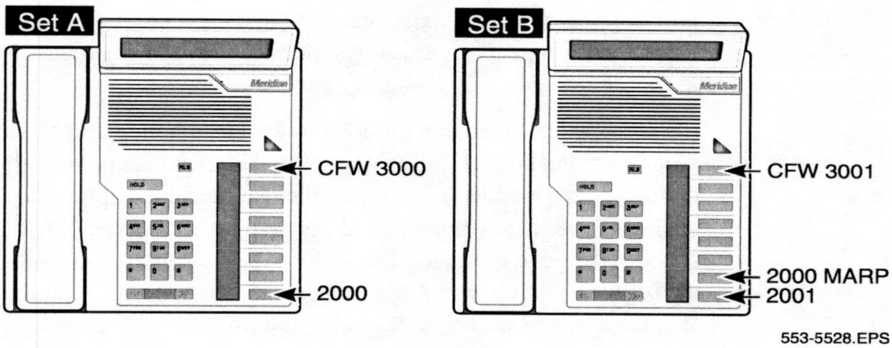
— If only Telephone A has CFW active, calls to DN 2000 are forwarded to DN 3000.

— If only Telephone B has CFW active, calls to DN 2000 are forwarded to DN 3001.

— If both Telephone A and B have CFW enabled, calls to DN 2000 are forwarded to DN 3000 because Telephone A is the MARP TN.

At times, even though the MARP TN is actually a secondary DN appearance, it can control where a call is redirected. Due to potential confusion, it is recommended that a secondary appearance not be defined as the MARP TN when a prime appearance is available. Refer to Figure 63.

Figure 63
MARP control



CFW DN on Telephone A is DN 3000. CFW DN on Telephone B is DN 3001.

— If both Telephone A and Telephone B have CFW active, all calls to DN 2000 go to DN 3001 because Telephone B is the MARP TN.

— If only Telephone A has CFW active, all calls to DN 2000 go to DN 3000.

— If only Telephone B has CFW active, no calls to DN 2000 are forwarded.

If all DN appearances are secondary, no calls are forwarded.

Call Forward No Answer

The MARP TN always controls the call redirection for Call Forward No Answer.

— If a DN is assigned as a Prime DN on a telephone and as a secondary DN on one or more telephones, the DN list is still organized as described in the preceding paragraphs. If only one prime appearance of a DN exists, however, call redirection parameters are derived from the TN of the prime appearance telephone, even though it may not be at the end of the list. A prime appearance is always the first TN used when the system looks for call redirection instructions.

— If a DN appears on analog (500/2500 type) telephones, and Meridian 1 proprietary telephones, the analog (500/2500 type) telephones are listed in numerical TN order at the top of the list. Meridian 1 proprietary telephones are listed in numerical TN order at the bottom of the list. A service change to an analog (500/2500 type) telephone moves its TN to the beginning of the list. A service change to a Meridian 1 proprietary telephone moves its TN to the end of the list.

— A SYSLOAD restructures the list back to numerical TN order with analog (500/2500 type) telephones at the top and Meridian 1 proprietary telephones at the bottom. Call redirection parameters continue to be derived as described in the preceding paragraphs.

Call Party Name Display

On ST and 21 systems, with X11 Release 17 and lower, the number of DN appearances restricts the number of letters/digits allowed for CPND. These engineering guidelines must be followed:

— 11 or fewer appearances allows 27 digits/letters in the name

— 12 appearances allows 23 digits/letters in the name

— 13 appearances allows 20 digits/letters in the name

— 14 appearances allows 16 digits/letters in the name

— 15 appearances allows 14 digits/letters in the name

— 16 appearances allows 11 digits/letters in the name

— 17 appearances allows 9 digits/letters in the name, and

— 18 appearances allows 8 digits/letters in the name.

Call Waiting Redirection

If the Multiple Appearance Directory Number Redirection Prime (MARP) feature is activated, the Call Forward No Answer (CFNA) treatment given by Call Waiting Redirection for an unanswered Call Waiting call follows the MARP feature for CFNA treatment of calls to an idle DN.

Hunting

The MARP TN always controls the call redirection for Hunting. Short Hunting takes precedence over Hunting and MARP. The MARP TN is referred to until Short Hunting is encountered. Short Hunting is in control until it expires. When short hunting expires, the MARP TN for the first DN in the Short Hunt sequence takes control.

Network Intercom

If more than one set is allocated the same prime DN, the Hot Type I call will terminate on the set designated as the Multiple Appearance Redirection Prime (MARP). If the MARP DN is not the prime DN on the set, or if the set designated as the MARP DN is not a Meridian 1 proprietary telephone, the first Meridian 1 proprietary telephone with the prime DN will be used. If none of these conditions are met, the call will terminate as a non-Hot Line call and the calling party will be notified on the display.

Hot Type D calls can have voice termination only on a MARP Terminal Number (TN), or if there is no MARP TN, then on the first TN in the TN list. A No Answer Indication for Hot Type D can only be left on the MARP TN, or if there is no MARP TN, then on the first TN in the TN list.

Phantom Terminal Numbers (TNs)

Multiple appearance and MARP cannot be enabled on a phantom TN.

User Selectable Call Redirection

When a Multiple Appearance DN is rung, the determination of the number of ringing cycles for CFNA depends on the value of the MARP prompt in LD 17. If the value is "YES," the number of ringing cycles is determined by the Ringing Cycle Option (RCO) number of the DN that is classified as a MARP TN. If the DN is a Multiple Appearance DN (MADN), the RCO values in the other TN blocks for that DN are ignored.

If the MARP value is "NO," the RCO is taken from the first TN in the DN block with a primary appearance of the DN. If there is none, the last TN in the DN block is used.

Feature packaging

Multiple Appearance Directory Number Redirection Prime is included in the base X11 system software.

Feature implementation

If MARP is not activated, the overlays listed have this message printed: "MARP NOT ACTIVATED." The message appears only once, at the very beginning of the overlay. When MARP is active, no message appears. The overlays are: LDs 10, 11, 20, 22, 25, 80, 81, 82, and 83.

When changing or adding a new Single Appearance DN to the system, the MARP TN is automatically assigned. The system indicates this TN is the MARP for the new DN with a MARP message.

When adding or changing a Multiple Appearance DN, the system indicates which TN is the current MARP TN. You can reassign the MARP TN if required.

SCH5524 appears at the end of the Service Change session, when the MARP TN has been changed.

LD 10 – Add an analog (500/2500 type) telephone with a Single Appearance DN.

Prompt	Response	Description
REQ	NEW	Add new data to the system.
TYPE	500	500/2500 telephone.
TN	l s c u c u	Terminal Number. For Option 11C.
DN	xxx...x	Directory Number.
- MARP		MARP prints on the next line indicating this TN is the MARP for DN xxxx.

LD 10 – Add an analog (500/2500 type) telephone with a Multiple Appearance DN.

Prompt	Response	Description
REQ	NEW	Add new data to the system.
TYPE	500	500/2500 telephone.
TN	l s c u c u	Terminal Number. For Option 11C.
DN	xxx...x	Directory Number.
- MARP ON TN	l s c u c u	*MARP ON TN l s c u* prints on the next line indicating TN l s c u (c u for Option 11C) is the current MARP.
- MARP	(NO) YES	(Do not) set the MARP to this new TN.
SCH5524DN nnnn NEW MARP	l s c u c u	This message indicates the MARP for the old DN nnnn is changed. The new MARP is TN l s c u (c u for Option 11C).

LD 10 – Changing a analog (500/2500 type) telephone with a Multiple Appearance DN.

Prompt	Response	Description
REQ	CHG	Modify existing data.
TYPE	500	500/2500 telephone.
TN	l s c u c u	Terminal Number. For Option 11C.
DN	xxx...x	Directory Number.
- MARP ON TN	l s c u c u	This message indicates the current MARP is TN l s c u (c u for Option 11C).
- MARP	(NO) YES	(Do not) set the MARP to this TN.
SCH5524DN nnnn NEW MARP	l s c u c u	The message indicates the MARP for the old DN nnnn is changed. The new MARP is TN l s c u (c u for Option 11C).

LD 11 – Add a telephone with a Single Appearance DN.

Prompt	Response	Description
REQ	NEW	Add new data to the system.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx aaa yyyy	xx is the key number aaa is the DN type: MCN (multi-call nonring) MCR (multi-call ring) SCN (single-call nonring), or SCR (single-call ring). yyyy is the DN.

- MARP		*MARP* prints on the next line indicating this TN is the MARP for DN yyyy.
KEY		Reprompts until <CR> is entered.

LD 11 – Add a telephone with a Multiple Appearance DN.

Prompt	Response	Description
REQ	NEW	Add new data to the system.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx aaa yyyy	xx is the key number. aaa is the DN type: MCN (multi-call nonring) MCR (multi-call ring) SCN (single-call nonring), or SCR (single-call ring). yyyy is an existing DN.
- MARP ON TN	l s c u c u	*MARP ON TN l s c u* prints on the next line indicating TN l s c u (c u for Option 11C) is the current MARP.
- MARP	(NO) YES	(Do not) set the MARP to this new TN.
KEY		Reprompts until <CR> is entered.
SCH5524DN nnnn NEW MARP	l s c u c u	This message indicates the MARP for the old DN nnnn is changed. The new MARP is TN l s c u (c u for Option 11C).

LD 11 – Changing a telephone with a Multiple Appearance DN.

Prompt	Response	Description
REQ	CHG	Modify existing data
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx aaa yyyy	xx is the key number. aaa is the DN type: MCN (multi-call nonring) MCR (multi-call ring) SCN (single-call nonring), or SCR (single-call ring). yyyy is the DN.
- MARP ON TN	l s c u c u	*MARP ON TN l s c u* prints on the next line indicating TN l s c u (c u for Option 11C) is the current MARP.
- MARP	(NO) YES	(Do not) set the MARP to the working TN.
KEY		Reprompts until <CR> is entered.
SCH5524DN nnnn NEW MARP	l s c u c u	This message indicates the MARP for the old DN nnnn is changed. The new MARP is TN l s c u (c u for Option 11C).

LD 10/LD 11 – Removing a MARP TN.

Prompt	Response	Description
REQ	OUT	Remove data from the system.
TYPE	aaaa	Telephone type, where: aaaa = 500, 2500, SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	TN l s c u (c u for Option 11C) is the MARP for DN nnnn. This is the TN that is being removed.
SCH5524DN nnnn NEW MARP	l s c u c u	This message indicates the MARP for the old DN nnnn is changed. The new MARP is TN l s c u (c u for Option 11C).

LD 17 – Activating or deactivating MARP.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PARM	Configuration Record. Release 19 gate opener.
PARM	YES	Change system parameters.
- MARP	YES NO	Activate or deactivate MARP. There is no default. <CR> retains the previous system data.

LD 20 or LD 22 – Print MARP information.

Prompt	Response	Description
REQ	PRT	Print information.
TYPE	TNB	Terminal Number data block.
	(DNB, SL1)	(Can also print out DN data block or telephone type.)

The printout will look like the following.

— For the DN data block:

DN 2000
TYPE SL1
TN 018 0 02 00 KEY 00 MARP DES NO DES NO DATE
TN 018 0 02 01 KEY 01 DES NO DES NO DATE

— For a telephone data block:

DES NO DES
TN 001 0 0 00
TYPE SL1
KEY 00 MCR 2000 MARP
01 MRK

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	All
Networking:	No

Multiple Console Operation

The Meridian 1 permits each customer to have up to 63 Attendant Consoles. X11 Release 7 and earlier software permits each customer to have up to 15 Attendant Consoles. Incoming calls are routed in a circular fashion to the first idle attendant. If all consoles are busy, calls are held in the attendant queue and are presented to the first idle attendant. Each console is identified by a customer-defined, two-digit Attendant Console number (01 to 63).

The assignment of Incoming Call Indicators (ICIs) and Trunk Group Busy (TGB) key/lamp pairs is identical for all Attendant Consoles in the customer group, except when Console Presentation Group Level Services, a multi-tenant feature, is configured. The flexible features key/lamp strip can be assigned on a per console basis.

The features that can be assigned to the flexible features strip include the following:

— Attendant Administration

— Autodial

— Automatic Wake Up

— Barge-In

— Busy Verify

— Call Park

— Calling Party Number

— Charge Account

— Controlled Class of Service, Enhanced

— Display Calls Waiting

— Display Date
— Display/Change Date
— Display Destination
— Display Source
— Display Time
— Display/Change Time
— Do Not Disturb (Individual)
— Do Not Disturb (Group)
— End-to-End Signaling
— Malicious Call Trace
— Message Cancellation
— Message Indication
— Mini-CDR Low Tape Alarm (the SL-1M only)
— Paging
— Routing Control
— Speed Call Controller
— System Speed Call Controller, and
— Stored Number Redial.

Operating parameters

Prior to X11 Release 8, only 15 Attendant Consoles per customer were permitted. X11 Release 8 and later software allows 63 consoles to be defined per customer.

Feature interactions

Departmental Listed Directory Number

Departmental Listed Directory Number (DLDN) supports the assignment of 63 consoles per DLDN.

Multi-Tenant Services

Up to 63 consoles can be defined in a single Console Presentation Group (CPG).

Feature packaging

Multiple Console Operation is included in basic X11 system software.

Feature implementation

The following overlays have been modified to allow input of 63 consoles on X11 Release 8 and later software:

— Attendant Console	LD 12
— Customer Data Block	LD 15
— Tenant-to-Tenant Access	LD 93

Feature operation

No specific operating procedures are required to use this feature.

Multiple Customer Operation

The Meridian 1 system can serve up to 32 (customer numbers 0-31) individual customers from the same machine. X11 Release 14 and later software supports 100 customer groups (numbered 0-99). Customers have their own features, restrictions, numbering plans, trunks, and special services. They are granted access to the system as if they are the sole user.

Operating parameters

There are none.

Feature interactions

System hardware, like Serial Data Interface (SDI), Digitone Receiver (DTR), Tone and Digit Switch (TDS), and Conference, is shared among all the customers on the machine.

The Speed Call list parameter (8191) applies to the machine, not the customer. It is shared among all customers on the system.

Feature packaging

Multiple Customer Operation (CUST) package 2 has no feature package dependencies.

Feature implementation

No change to existing configuration is required to implement the Multiple Customer Operation feature.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	21
Networking:	No

Multiple Queue Assignment

The Multiple Queue Assignment (MQA) feature enhances the capabilities of Automatic Call Distribution (ACD). This feature allows agents to service multiple queues simultaneously.

MQA will allow Call Center customers to achieve a high level of control over the manner in which agents are assigned to calls in queues (ACD-DNs). Thus, Call Center managers will be able to better direct calls to agents whose skills best meet the needs of the caller. This feature will also allow agents to login to up to five queues, select their priorities within these queues, and choose their supervisor's position ID. Agents will also be able to login at any ACD set and have their non-ACD calls forwarded to the set into which they are logged.

Refer to *ACD Description , Operations and Tests* (553-2671-110) for further details.

Multi-purpose Serial Data Link D-Channel Handler

This feature introduces the Multi-purpose Serial Data Link (MSDL) D-Channel Handler into international markets. The capabilities provided for international applications are identical to those described in the Northern Telecom Publication (NTP) ***Multi-purpose Serial Data Link***.

Introduced in X11 Release:	19
Networking:	Yes

Multi-purpose Serial Data Link Serial Data Interface

Serial Data Interface (SDI) is supported by the Multi-Purpose Serial Data Link (MSDL) card with X11 Release 19 and later. SDI extends the input/output capability of the MSDL card by providing an asynchronous serial data interface. SDI is composed of software components that reside on the Meridian 1 and the MSDL.

For a complete description of MSDL SDI, please refer to *X11 system management applications* (553-3001-301).

Introduced in X11 Release:	7
Networking:	No

Multi-Tenant Service

Multi-Tenant Service feature allows Meridian 1 customers to resell Meridian 1 features and services to other users. The stations belonging to the customer can be divided into customer sub-groups known as tenants. Tenants are separated by programming access restrictions on a tenant by tenant basis.

Access to other tenants, Attendant Consoles and trunk routes can be configured so that tenants can private use of some services and shared use of some services. As well, Multi-Tenant Service can also be configured to denied access to service. Records tenant activity is maintained by Call Detail Recording (CDR).

Telephones that are not assigned tenant status belong to the Meridian 1 customer. These customer resource telephones have access to all other telephones, Attendant Consoles and outgoing trunk routes belonging to the customer.

The number of tenants that can be configured on a per customer basis is dependant on the number of configured customers and the amount of available memory. The maximum number of tenants is 512 per customer.

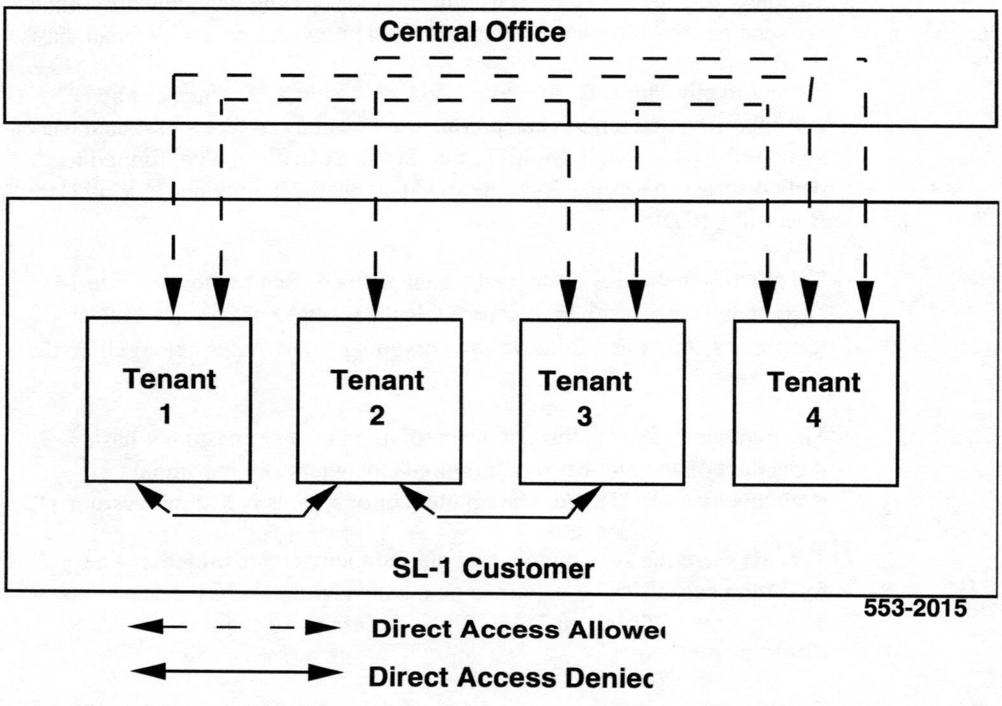
Tenants share the same numbering plan of their service provider. The following capabilities are defined on a tenant by tenant basis: Tenant to Tenant Access, Tenant to Trunk Route Access and Tenant to Attendant Console Grouping.

Tenants receive all the features defined by the Meridian 1 customer. Features that are handled at the tenant level include Incoming Call Indicators, Call Waiting Indicator, Recorded Overflow Announcement, Listed Directory Numbers, Attendant Overflow Position and Night Directory Number.

Tenant-to-Tenant Access

Calls between tenant groups for the same customer are defined by Tenant-to-Tenant Access. As shown in Figure 64, tenant is configured to allow direct internal call access to some or all tenants of the same customer. Likewise, a tenant can be denied direct access to other tenants. To reach these tenants, the caller must dial the tenant's Listed Directory Number. Access is always two-way. Therefore, if Tenant A has direct internal call access to Tenant B, Tenant B also has the same access to Tenant A. Customer telephones not belonging to a tenant have two-way access to all tenant telephones in the customer group.

Figure 64
Tenant-to-Tenant access



As shown in Table 110 Tenant-to-Tenant Access allows or denies tenants of the customer:

Table 110
Tenant-to -Tenant Access allowed or denied

Tenant	Direct access allowed	Direct access denied
1	2	3 & 4
2	1 & 3	4
3	2	1 & 4
4		1,2 & 3

Outgoing Tenant-to-Trunk Route Access

Tenant access applies only to outgoing calls. All tenants have access to incoming calls on any route. Customer telephones have access to all the customer's outgoing routes.

A tenant can have private outgoing trunk routes assigned. This is done by denying all other tenants access to the routes. Figure 65 shows examples of the following.

Table 111
Tenant Access to Private Routes

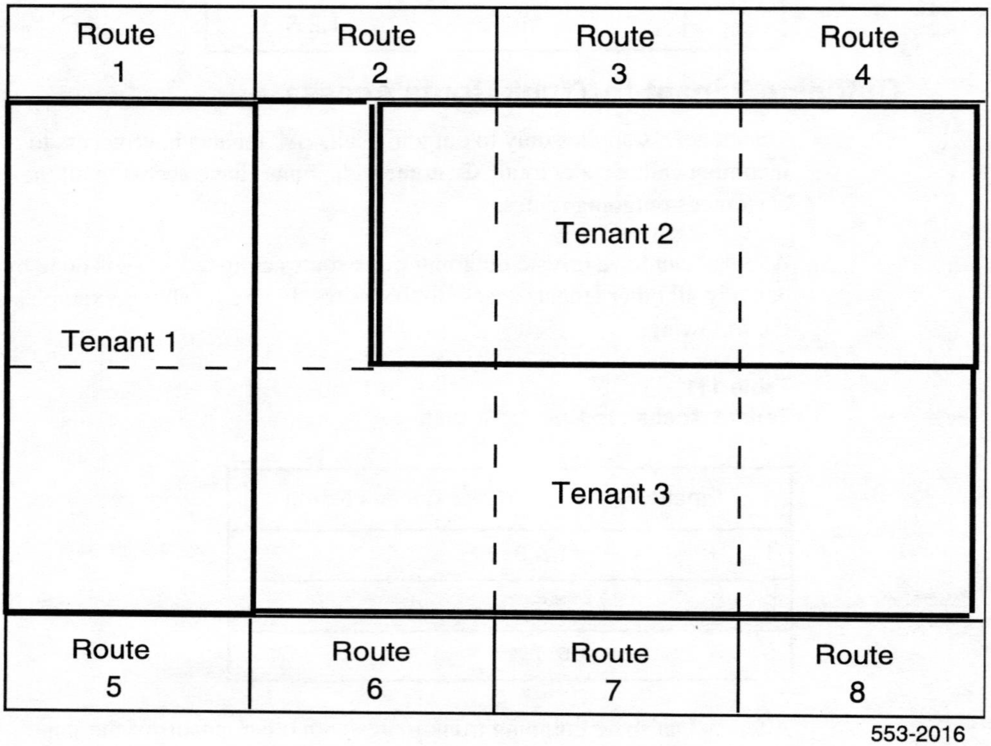
Tenant	Private Access Route
1	1 & 5
2	3 &4
3	6, 7 & 8

A tenant can share outgoing trunk routes with other tenants of the same customer. As shown in Figure 65, Tenants 2 and 3 share access to route 2.

Table 112
Tenant Restriction s to Outgoing Routes

Tenant	Restricted Access to Trunk Routes
1	2, 3, 4, 6, 7 & 8
2	1, 5, 6, 7 & 8
3	1, 3, 4 & 5

Figure 65
Tenant-to-Trunk Route Access



Attendant Console Groups

With Multi-Tenant Service, all Attendant Consoles are placed into groups which are associated with specific tenants and specific incoming trunk routes. The Group Number range is 0 to 511. All Attendant Consoles configured for a customer are automatically members of group 0. The other groups are defined in the software to fit tenant requirements. [Refer to the section on Functions in this document for a description of the structure and functions of the Attendant Console Groups.]

Tenant-to-Attendant Access (Internal Calls)

Tenant-to-Attendant Access specifies which Attendant Console Group receives automatic presentation of a tenant's dial-zero calls.

Trunk Route-to-Attendant Access

Route-to-Attendant access specifies which Attendant Console Group receives automatic presentation of incoming calls from a particular route.

Console Presentation Groups

Console Presentation Groups (CPGs) are assigned to handle attendant calls from one tenant for a customer or for calls originating from certain trunks in a particular route.

Most of the Attendant Console features and parameters apply to CPGs. For a complete description of the functionality for CPGs, please refer to the section on CPGs in this document.

Attendant Console

Internal attendant-DN calls

When a tenant telephone dials the attendant DN, the call is presented to an idle Attendant Console. The call is routed to an Attendant Group associated with the tenant of the calling telephone, if Attendant Console Groups have been specified for the tenant. Otherwise, calls are presented to any idle Attendant Console belonging to the customer. For example, in Table 113, an attendant DN call from a Tenant 2 telephone is presented to an idle attendant in group 2 (consoles 1 or 2).

Incoming external calls

Incoming external calls are presented only to the Attendant Console Group specified to serve the trunk group. Also from Table 113, incoming calls on route 3 are presented to Attendant Consoles in group 6 (consoles 9 or 10).

Attendant Initiated Calls

All attendants have access to the customer's numbering plan and can initiate calls to any of a customer's tenants.

Attendant Overflow Position (AOP)

The Attendant Overflow Directory Number (AODN) should be accessible to all tenants. Attendant calls from tenants who do not have AODN access will not divert to AODN. They remain in the attendant queue.

Attendant Recall

When a tenant telephone recalls the attendant, the call is presented to an attendant in a group specified for the tenant of the calling telephone.

Attendant Extended Call

When an attendant extends a call from tenant A to tenant B, a 3-way conversation is set up only if tenant A and tenant B are allowed Tenant-to-Tenant Access.

Automatic Timed Recall (ATR)

When Automatic Timed Recall (ATR) alerts the attendant, the call is presented to an attendant within the Tenant group of the originally called number.

Console Presentation Group

A Console Presentation Group (CPG) is a subset of the consoles configured for a customer. A CPG is assigned to handle attendant calls from one tenant for a customer. A CPG can also be assigned to handle calls originated by trunks on a route.

CPG improves functions on the following CPG Level Services:

— **Attendant Overflow Positions** Each CPG can have its own AOP-DN and waiting time threshold specified.

— **Call Waiting Indication** The count thresholds, timers and buzz options for Call Waiting are defined for each CPG.

— **Incoming Call Identification** The ICI keys are defined for each CPG. Attendants see only those ICI key definitions for their own CPG.

— **Listed Directory Numbers** Each CPG allows four (4) LDNs.

— **Night Service** Each CPG can go into Night Service mode regardless of the status of the other CPGs.

Access to incoming trunk routes

Any tenant can access on an incoming call from any incoming trunk route. Attendant Console Groups can be specified to receive automatic presentation of incoming calls from specified routes. This includes calls that terminate at an Attendant Console and calls that intercept to an Attendant Console as well. For example, as seen in Table 113, incoming calls on route 2 are automatically presented to Attendant Console Group 5 (console 7 only).

Table 113
Typical attendant group arrangement

Attendant group number	Attendant consoles	Incoming Trunk routes	Tenant
0	1-10		
1	1		1
2	1, 2	1	2
3	1		3
4	3, 4	4	
5	7	2, 5	
6	9, 10	3	

Access to outgoing trunk routes

Tenants dial the appropriate trunk route Access Code to connect to a trunk route. Access Codes are assigned on a trunk route basis. Therefore, all tenants use the same Access Code to connect to a particular route. Customer telephones have access to all outgoing trunk routes belonging to their customer. Access to specific trunk routes is allowed or denied to individual tenants through service change. Tenants who attempt to access denied routes receive normal intercept treatment.

Operating parameters

Multi-Tenant Service is not supported by Meridian Mail applications.

Traffic data is collected on a per customer basis only.

Tenants may have private or shared access to the Modem Trunk routes configured for their customer.

All tenants have access to their customer's Music trunks.

Tenants may have private or shared access to the Paging routes configured for their customer.

All tenants have access to their customer's recorded Announcement (RAN) trunks.

Individual tenants can be allowed or denied trunk access (private or shared) for the following trunk types: Add-on Data Module, Centralized Automatic Message Accounting, Common Controlled Switching Arrangement, Central Office, Direct Inward Dialing, Dictation trunk, Direct Outward Dialing, Foreign Exchange, Modem, Paging trunk, TIE, Wide Area Telephone Service.

There are no restrictions on calls which are routed to the following trunk types: Automatic Identification of Outward Dialing, Music trunk, Recorded Announcement, Release Link, Main, Release Link, Remote Emergency Recorder

Feature interactions

Access restrictions

Multi-Tenant Access restrictions affect the way that tenants interact with other tenants, trunk routes and Attendant Consoles.

In general, Multi-Tenant Access restrictions take precedence over the Meridian 1 features with which they interact.

For example, when a direct Tenant-to-Tenant call has been made, the called party cannot transfer the call to a different tenant if the first and third tenants are denied access to each other.

In addition to Class of Service and Trunk Group Access Restrictions (TGAR)/Trunk Access Restriction Groups (TARG) restrictions, Multi-Tenant Service may impose the following access restrictions:

— Tenant-to-Tenant,

— Tenant-to-Trunk Group,

— Tenant-to-Attendant Group, and

— Trunk Group-to-Attendant Group.

Attendant Administration

An Attendant can dial the access code and activate the Administration Mode for that CPG group. In this mode, attendants can modify the configuration of any set for this customer.

Internal calls

When the caller's tenant has access to the tenant of the night telephone or trunk route, calls are delivered to the ACD Night Call Forward DN. When the caller's tenant does not have access to the tenant of the night telephone or trunk route, callers receive normal intercept treatment.

Automatic Timed Recall

When Automatic Timed Recall (ATR) alerts the attendant and Multi-Tenant Services are in effect, the call is presented to an attendant in the same tenant group as the originally dialed DN.

Basic Authorization Codes

All tenants share their customer's Authorization Code tables; however, Tenant-to-Tenant and Tenant-to-Trunk Route specifications override Basic Authorization Codes (BAUT).

Call Detail Recording

With the Multi-Tenant Service, all tenants are included in CDR records. The tenant numbers of the originating and terminating parties are added to the CDR records as shown in Table 114.

Table 114
CDR record types and descriptions

CDR record type	Description
A	Authorization Code
C	Charge Account
E	End
L	Internal Record
M	Charge Conference
N	Normal
P	Calling Party Number
Q	Connect Record
S	Start

Tenant and customer numbers are included by the system in the CDR output to provide the customer with data for call billing and chargeback activities.

Call Forward All Calls

Originating Party COS

If the calling party (CFO) option is defined in the Customer Data Block (LD 15), inter-tenant Call Forward is allowed if the calling party's tenant has access to the tenant of the Call Forward DN as well as access to the tenant of the dialed DN. If the Call Forward DN is in a Tenant group inaccessible to the caller, the DN is treated as invalid, and the overflow tone is returned to the caller. An access check is done by the software.

Forwarding Party COS

If the forwarding party (CFF) option is defined in the Customer Data Block (LD 15), inter-tenant Call Forward is allowed if the Call Forwarding party's tenant has access to the tenant of the Call Forward DN. The local Telephone Company decides whether the option is defined.

Call Forward Busy

DID calls to a busy telephone are forwarded to an idle Attendant Console specified for the tenant of the dialed telephone.

Hunting and Call Waiting take precedence over Call Forward Busy.

Call Forward No Answer

Attendant option

After a customer defined number of rings, an unanswered call is forwarded to an idle Attendant Console specified for the tenant of the dialed telephone.

Any DN option

If the tenant of the calling party has access to the tenant of the Call Forward DN, the unanswered call is forwarded to the Call Forward DN. If Tenant-to-Tenant Access is denied, the call is processed as if no CFNA-DN existed.

Secretarial Filtering

Calls receive Secretarial Filtering only if the tenant of the Call Forward DN is accessible by the tenant of the caller.

Call Forward No Answer, Second Level

All of the same operations apply to the forwarded DN with Second Level CFNA allowed.

Call Forward by Call Type

The originally dialed DN must have access to the tenant of the forwarding DN. This allows external calls to be easily forwarded to the programmed DN.

If the system is forwarding an internal call by CFCT, the originator must have access to the tenant of the programmed forwarding DN.

Call Park

Parked calls recall back to the Attendant who parked them. If that attendant goes into Position Busy mode, then the Parked call recalls to an attendant in the same CPG as the original. Recalls to Attendants going into Night Service mode are returned to the attendant queue until the caller abandons the call.

Call Transfer

A telephone user can transfer its original party to a third party only if the transferred parties can access each other. Software prevents joining tenants who are denied access to each other.

Calls Waiting Indication

The Calls Waiting Indication displays the calls waiting count for the customer. It is not tenant related, but because routes and tenants specify the consoles to which calls are automatically presented, it is possible that a non-zero call waiting count will be displayed. This occurs even though no calls are being presented to the console.

Centralized Attendant Service

Specific Attendant Consoles can be assigned to receive automatic presentation of incoming calls from Release Link-Main (RLM) trunks.

All tenants have access to Release Link-Remote (RLR) trunks.

Code restriction

The code restriction data configured for a customer, applies to all tenants belonging to that customer.

Conference

All members of a conference must have access to each other. Meridian 1 software runs an access check which prevents the addition of access denied tenants.

Controlled Class of Service

The tenant of the Controlled Class of Service Controlling Station must have access to the tenant of the controlled telephone in order to activate CCOS.

Departmental Listed DN

The Departmental Listed Directory Number (DLDN) takes precedence over Multi-Tenant Service. For either Dial-Zero or Recall, initiated from a tenant telephone, two events may occur. First, the call is presented to the DLDN attendant when the telephone has specified DLDN. Second, the call is presented to the console specified by the telephone's tenant when the telephone does not have DLDN specified.

Dial Intercom Group

The tenant of the dialing telephone must have access to the tenant of each telephone reached by Dial Intercom Group (DIG) dialing.

Electronic Switched Network

All tenants have access to the Electronic Switched Network (ESN) features specified at the customer level. Except for Tenant-to-Route access, all ESN features are identical for each tenant belonging to the same customer.

Coordinated Dialing Plan

All tenants are allowed access to all of a Coordinated Dialing Plan (CDP) if they are configured for access to TIE trunk routes that are a part of the CDP.

Flexible Call Back Queuing

The originating tenant must have access to an eligible route in the Call Back Queue (CBQ) route list.

Free Calling Area Screening

Free Calling Area Screening checks occur normally if the originating tenant has access to the selected route.

Basic Alternate Route Selection
Network Alternate Route Selection

All tenants have access to the BARS/NARS Access Codes of their customer. Tenants that do not share access to the selected route are denied access to that route.

Network Authorization Code

Network Authorization Code (NAUT) does not override Tenant-to-Route Access restrictions within the call originator's Meridian 1.

Network Speed Call (NSC)

All tenants have access to their customer's NSC lists. Any route selected via NSC must have Tenant-to-Route Access allowed.

Off-Hook Queuing (OHQ)

OHQ is allowed if the tenant has access to a route in the initial route list of their customer that is eligible for OHQ.

Flexible Hot Line

Flexible Hot Line allows designated telephones to place calls to a predetermined destination by going off hook. The Hot Line telephone's tenant must have access to the tenant of the Hot Line DN, or standard intercept treatment is provided.

Group Call

Group Call allows a QSU telephone user to place a call to a maximum of 10 (maximum of 6 for Option 11) predefined DNs simultaneously by pressing a Group Call key. The tenant of the telephone initiating the Group Call must have access to the tenant of each member in the group. Restricted members are excluded from the group. The Meridian 1undertakes access checks the originator against each group member.

Hunting

Circular, Linear, Secretarial or Short Hunting routes call from a busy DN to the next idle DN in a prearranged group. If the DN being hunted is not accessible by the dialing telephone, it is handled as an invalid member in the hunting chain. Short Hunting requires that all DNs configured on a QSU telephone belong to the same tenant.

Hunting Route

One step Route Hunting takes place between routes of the same trunk type. Tenants share their customer's route hunting specification and can use the stepped to route if they have Tenant-to-Route Access allowed for the route.

Integrated Messaging System (IMS)

Tenants can share or be denied access to their customer's IMS.

Integrated Voice Messaging System (IVMS)

Tenants can share or be denied access to their customer's Integrated Voice Messaging Service (IVMS). Tenants who do not have direct access to each other can use the IVMS Broadcast capability to leave messages for each other.

Intercept Treatment

All tenants share the customer's intercept specification.

When Tenant-to-Route Access restricts a Basic Alternate Route Selection (BARS)/Network Alternate Route Selection (NARS) call, intercept treatment is the same as any invalid BARS/NARS call.

When an internal call intercepts to an attendant because of defined restrictions or dialing irregularities, it automatically presents the call to one of the Attendant Consoles specified for the calling tenant.

When intercept treatment includes a Recorded Announcement (RN) and Tenant-to-Tenant Access restricts a call, an Access Denied RAN plays.

Field Lamp Array

The Lamp Field Array, located on either an Attendant Console or a QSU telephone, indicates the busy/idle status of 150 consecutive DNs. These DNs are displayed regardless of Tenant-to-Tenant Access specifications of the array equipped tenant telephone. For this reason, the DNs assigned in the array should be accessible by the tenant of the array associated telephone.

Maintenance telephone

QSU telephones with maintenance allowed COS must be allowed access to all tenants, all trunk routes and all Attendant Consoles.

Manual service

When a manual telephone goes off-hook, the call is presented to an idle Attendant Console belonging to a group specified for its tenant.

Manual Trunk service

When an incoming trunk terminates on a DN, there is no access check. Incoming trunks terminate on an Attendant Console only if the console is specified for that manual trunk route.

Tenant-to-Route access checking is completed for outgoing manual trunk calls.

Multiple Appearance DNs

All appearances of a DN should reside on telephones belonging to the same tenant. When a multiple appearance DN is called, the last non-fully restricted Terminal Number (TN) in its TN list determines the terminating tenant number for Tenant-to-Tenant Access checking.

Multiple Listed Directory Numbers

Route-to-Attendant Console Access determines which Attendant Console Group receives automatic presentation of calls from a specific direct inward dialing (DID) trunk route. Each of the four DID LDNs are configured to have its calls presented at the loop key of specific Attendant Consoles by using DLDN.

Night Service

Automatic Call Distribution (ACD) allows special functionality for the system under certain conditions, such as Night Service.

The Night DN should be assigned as a customer resource so that when Night Service is in effect, all tenants have access to the Night DN for internal calls. Otherwise the call is treated as if no Night DN exists.

Position Busy

When all Attendant Consoles designated to receive incoming trunk calls from a particular trunk route are in Position Busy, incoming trunk calls from those routes are directed to the Trunk Night Service DN.

Office Data Administration System

Office Data Administration System (ODAS) does not contain tenant information.

Ring Again

Ring Again is permitted when the originating tenant has access to the destination tenant.

Ringing Number Pickup

Ringing Number Pickup (RNPU) enables a telephone to answer calls to other telephones in the same RNPU Group. All tenants have access to their customer's RNPU Access Code. Members of an RNPU group can only answer calls for other members if their tenant has access to the tenant of the calling party. For this reason, members of an RNPU group are selected from telephones belonging to the same tenant. The calling party's access is checked against the called party by the Meridian 1.

Route Selection-Automatic Number Identification

All tenants can dial the Route Selection - Automatic Number Identification (RS-ANI) DN. The ANI route selected from the RS-ANI list is used only if the tenant of the originating telephone has access to the route.

Secrecy

The Secrecy option, specified for a customer, applies to all CPG attendants for that customer.

Speed Call

Speed Call allows a telephone user to place calls to specified DNs by dialing a two-digit code. A user of a Speed Call List receives normal intercept treatment if the tenant does not have access to the listed destination tenant.

Supervisory consoles

Supervisory consoles specified for a customer belong to one Console Presentation Group (CPG). In the Supervisory mode, ICI lamps show only the information for ICIs in that CPG. The thresholds specified in the Customer Data Block apply only to the CPG where that console resides, and they do not effect any other CPG.

System Speed Call

All tenants share their customer's System Speed Call (SSC) lists. When a System Speed Call DN is used Tenant-to-Trunk Route access restrictions apply.

Trunk Group Access Restrictions

All tenants share their customer's Trunk Group Access Restrictions (TGAR), but Tenant Service Access restrictions take precedence, even though the telephone COS and TGAR do not restrict access to a route. Normal intercept treatment is provided when Tenant Service Access is denied.

Trunk routes
Voice Call

Tenant-to-Tenant Access must be allowed between the Voice Call originating telephone and terminating telephone.

Feature Packaging

The following packages are required for Multi-Tenant Service:

— Multi-Tenant Service (TENS) is package 86, which requires:

- Console Presentation Groups (CPGS) package 172.

Other features expected in a Console Presentation Group environment must be packaged for complete functionality. They are as follows:

— Centralized Attendant Service-Remote (CASR) package 26;

— Centralized Attendant Service-Main (CASM) package 27;

— Recorded Overflow Announcement (ROA) package 36;

— Attendant Overflow Position (AOP) package 56;

— The maximum number of route list entries for BARS/NARS is always 64, independent of the CPG Level Services package (172); and

— CPG services are mutually exclusive with Departmentally Listed DNs (DLDN).

Feature implementation

LD 93 – Enable, disable, or print Multi-Tenant Service for a specified customer. Ensure that the customer night DN and the attendant overflow DN (if assigned) are accessible by all tenants.

Prompt	Response	Description
REQ	NEW, OUT, PRT	Add, remove, or print.
TYPE	TENS	Tenant service data block.
CUST	xx	Customer Number.

LD 93 – Allow, deny, or print Tenant-to-Tenant Access for a specified tenant.

Prompt	Response	Description
REQ	CHG, PRT	Change, or print.
TYPE	TACC	Tenant-to-Tenant Access Data Block.
CUST	xx	Customer Number.
TEN	1-511	Tenant number. Tenant 0 is reserved for telephones with a TEND Class of Service.
ACC	DENY ALOW	Tenants denied access are to be entered. Tenants allowed access are to be entered.
DENY	1-511 1-511 ALL	Tenant numbers denied access to and from this tenant (prompted if ACC = DENY). All tenant numbers denied access to and from this tenant (tenant can only access itself).
ALOW	1-511 1-511 ALL	Tenant numbers allowed access to and from this tenant (prompted if ACC = ALOW). All tenant numbers allowed access to and from this tenant.

LD 93 – Allow, deny, or print Tenant-to-Route Access for a specified trunk route.

Prompt	Response	Description
REQ	CHG, PRT	Change, or print.
TYPE	RACC	Tenant-to-Route Access Data Block.
CUST	xx	Customer Number.
ROUT	0-511	Route Number.
ACC	DENY ALOW	Tenants denied access to the route are to be entered. Tenants allowed access to the route are to be entered.
DENY	1-511 1-511 ALL	Tenant numbers denied access to this route (prompted if ACC = DENY). All tenant numbers denied access to this route.
ALOW	1-511 1-511 ALL	Tenant numbers allowed access to this route (prompted if ACC = ALOW). All tenant numbers allowed access to this route

LD 93 – Add Attendant Console group.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	ACG, CPG	Attendant Console group data block.
CUST	xx	Customer Number.
AGNO	1-63	Attendant Console group number. Attendant Console group 0 (AGNO 0) always exists and contains all Attendant Consoles configured for the customer.
ANUM	1-63 1-63	Add attendant console numbers (before X11 Release 8, only 15 Attendant Consoles are allowed).

LD 93 – Assign Tenant-to-Attendant Console access.

Prompt	Response	Description
REQ	CHG, PRT	Change, or print.
TYPE	TACG, TCPG	Tenant-to-Attendant Console access data block.
CUST	xx	Customer Number.
TEN	1-511	Tenant number. Tenant 0 is reserved for telephones with a TEND Class of Service.
AGNO	0-63	Attendant Console group number.

LD 93 – Assign Attendant Console access.

Prompt	Response	Description
REQ	CHG, PRT	Change, or print.
TYPE	RACG, RCPG	Route-to-Attendant Console access data block.
CUST	xx	Customer Number.
ROUT	0-511	Route Number.
AGNO	0-63	Attendant Console group number.

LD 10 – Add Multi-Tenant Service assignments on Analog (500/2500) telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11.

CLS	(TENA) TEND	Tenant service (allowed). Tenant service denied (station shares customer resources and is a non-tenant).
TEN	1-511	Tenant number (prompted if CLS = TENA). Tenant 0 is reserved for telephones with a TEND Class of Service.

LD 11 – Add Multi-Tenant Service assignments on Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11.
CLS	(TENA) TEND	Tenant service (allowed). Tenant service denied (station shares customer resources and is a non-tenant).
TEN	1–511	Tenant number. Tenant 0 is reserved for telephones with a TEND Class of Service. Prompted if CLS = TENA.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	19
Networking:	No

Multi-User Login

Meridian 1 Multi-User Login (MULTI_USER) package 242 enables up to three users to log in, load, and execute overlays simultaneously. These three users are in addition to an Attendant Console or maintenance terminal. The Multi-User Login capability increases the efficiency of technicians by enabling them to perform tasks in parallel.

For a complete description of Multi-User Login, please refer to *X11 system management applications* (553-3001-301).

Music

The Music Package supports Music on Hold and Automatic Call Distribution (ACD) Music on Delay. One or more music sources can be connected to one or more music trunks on peripheral equipment. Each music trunk is assigned to a music route and to a conference loop. Incoming callers are bridged into a listen-only conference and provided with music when on hold or when waiting for an ACD call to be answered.

Music on Delay

Music on Delay presents a listen-only path to a music source for calls waiting in ACD queues. Music on Delay sources are identified separately for each Automatic Call Distribution Directory Number (ACD DN). Complete details are described in *Automatic Call Distribution feature description* (553-2671-110).

Music on Hold

This feature allows incoming calls over a CO, FX, WATS, DID, or TIE trunk to receive music if placed on hold. Music is provided only if the trunk route is defined to receive music. The trunks selected to receive music are provided with a listen-only path to a music conference connection.

Music is provided by a dedicated music trunk by means of the conference circuit. To minimize blocking of the music conference, at least two conference loops must be assigned in each network group requiring music. The loop with the higher number should not be assigned to music trunks.

Operating parameters

Music is provided by a Recorded Announcement (RAN) or universal trunk circuit card.

Only trunks assigned to a route specified by service change receive Music on Hold.

When a call is held, the system looks for a network path to provide the music. If a path is not found, no music is heard.

When a Universal trunk card is used, Music and RAN trunks can be assigned to the same card.

Connections blocked once are not automatically attempted again.

Simple source-only connections on the Attendant Console receive music; all others do not.

Main Release Link Trunks do not receive music.

Calls to special trunks (such as Paging or Dictation) do not receive music if placed on hold.

The music trunk Terminal Number (TN) must be within the same network group as the conference circuit to which it is assigned.

One music trunk per customer must be located in each network group requiring music.

Music is not supplied across groups (if group 4 does not have a music trunk and groups 0-3 have music trunks, an incoming call to group 4 placed on hold will not receive music).

A single conference loop with one music trunk assigned can support up to 29 simultaneous listeners.

If more than one music trunk is assigned to one conference loop, they must use different routes. The total number of possible listeners is 30 minus the number of assigned trunks. Additional music trunks and conference loops can be configured if required.

The music source must be compatible with the music trunk circuit pack.

Feature interactions

AC15 Recall: Transfer from Norstar

A party put on hold by an AC15 trunk will hear music if it is configured.

Attendant Trunk Group Busy Indication

A music route that appears on a Trunk Group Busy key on the Attendant Console cannot be controlled by activation of the Trunk Group Busy key. In addition, the associated lamp will not reflect the status of the music trunks.

Break In with Secrecy

During secrecy, if there is only one undesired party in the conference, music is not provided to this party when excluded. However, intrusion tone is given to this party.

Call Park

When a call is parked, music is not heard. When a trunk is parked, music plays if music is enabled for the route.

Conference

With basic Music on Hold, when a call is placed on consultation hold while a Conference is being established, music does not play. Enhanced Music (EMUS) package 119 is required for music on consultation hold (see "Music, Enhanced" on page 2061).

Group Hunting Queuing Limitation

No music is provided for Group Hunting Queuing Limitation.

On Hold on Loudspeaker

Music on Hold is not be heard by either party during a loudspeaker call.

Recovery on Misoperation of Attendant Console

Music on Hold is applied to calls put on hold due to AUTOHOLD.

Source Included when Attendant Dials

The source is included in a conference involving the attendant, the source, and Recorded Announcement or music treatment. Intrusion tone is not provided in this case.

Trunk Traffic Reporting Enhancement

The Trunk Seizure Option is not supported on a music trunk.

Feature packaging

Music (MUS) package 44 requires:

— Recorded Announcement (RAN) package 7.

Feature implementation

LD 17 – Add or change conference loops for Music on Hold.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN CEQU	Configuration Record. Release 19 gate opener.
CEQU	(NO) YES	Change to CE parameters.
- XCT	0-158	Loop number for NT8D17 Conference/TDS/MFS card. Enter an even network loop number for TDS/MFS functions. The conference function is automatically assigned the next higher (odd) loop number.
- CONF	0-158	Loop number for conference card.

LD 16 – Add or change a music route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
CUST	0-99 0-31	Customer number. For Option 11C.
TKTP	MUS	Music route.
ICOG	OGT	Outgoing route only.
ACOD	xxxx	Trunk route access code.

Note: All other prompts can be set to default values.

LD 14 – Add or change a music trunk.

Prompt	Response	Description
REQ	NEW CHG	New or change.
TYPE	MUS	Music trunk.
TN	l s c u c u	Terminal Number. For Option 11C.
CUST	0-99 0-31	Customer number. For Option 11C.
RTMB	xxx yyy	Route number and member number.
CFLP	0-158	Conference loop assigned to music in LD 17.

LD 16 – Enable/disable Music on Hold for trunk routes.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
CUST	0-99 0-31	Customer number. For Option 11C.
ROUT	0-511 0-127	Route number. For Option 11C.
TKTP	COT DID FEX TIE WAT	Route type.
MUS	(NO) YES	Music on Hold (is not) or is to be provided for this trunk route.
MRT	xxx	Music route number.

LD 23 – Enable/disable Music for an Automatic Call Distribution Directory Number.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	ACD	Update the ACD data block.
CUST	0-99 0-31	Customer number. For Option 11C.
ACDN	xxx...x	ACD DN.
MURT	X 0-511	Music route number. X = remove route.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	23.0x
Networking:	No

Music Broadcast

The Music Broadcast feature expands existing Music functionality. This feature provides the following enhancements:

— Broadcast Capabilities

— Incremental Software Management limit

— Traffic Study Option

Broadcast Capabilities

Existing Conference-based Music features require that each Music trunk be assigned to a Music route and to a Conference loop. Incoming callers are bridged into a listen-only conference and provided with music while on call hold or call waiting in an Automatic Call Distribution (ACD) environment. The existing Conference-based Music features support intra-group music only. Therefore, each network group must be provided with its own Music trunk.

The Music Broadcast feature allows the Meridian 1 system to broadcast music to several parties at one time via a single Music Broadcast trunk port. This feature supports Music on Hold (MOH). Music is now delivered via X11 software; hence, Conference hardware is not required. It is no longer necessary to share Conference resources with Conference features, such as Conference and Group Call. Music Broadcast supports both intra-group and inter-group music. Therefore, a Music trunk in each network group is not required.

A Music Broadcast call consists of several one-way connections from the Music trunk to each caller. The Music Broadcast feature reduces the number of timeslots required for callers to listen to music while on call hold or call waiting in an Automatic Call Distribution (ACD) environment. One timeslot is required to enable Music trunk broadcasts. In addition, each party listening to music through the broadcasting music trunk requires one broadcast connection. The extra speech path resources that are needed for the existing Conference-based Music are unnecessary for Music Broadcast.

Incremental Software Management

An Incremental Software Management (ISM) limit is introduced for the Music Broadcast feature. This limits the total number of Music Broadcast connections allowed on a system. The ISM limit can be allocated over different Music routes and trunks. ISM allows a total of 64 Music Broadcast connections on one trunk at one time. If one trunk is configured with 64 connections, when the limit is reached, the 65th caller hears silence. The Music trunk is no longer available until a call disconnects. When a call disconnects, the next caller receives one of the Music Broadcast connections and receives music. However, the 65th caller still hears silence, even though a connection has become available.

If a customer has 64 connections configured on one trunk but requires more connections, additional trunks can be added to their system and additional connections can be purchased incrementally to a maximum of 9,999 connections. For example, should this customer require a total of 124 connections, an additional trunk and an additional 60 connections can be added to their original configuration. This provides the customer with a total of 124 connections (64 connections + 60 connections). Overlay 22 is modified to print the new ISM information for Music Broadcast connections. The existing SLT command prints the ISM information for the system.

The existing ISM header in Overlay 14 is modified to indicate the number of Music Broadcast connections allowed for the system. AVAIL shows the system's ISM limit for Music Broadcast connections. USED shows the number of configured Music Broadcast connections (the total number of Music Broadcast trunks for the system multiplied by the maximum number of connections per trunk). TOT shows the maximum number of Music Broadcast connections that can be supported on one system (AVAIL + USED).

The existing TN information shown in the ISM header in Overlay 14 is not modified by the Music Broadcast feature, as the amount of Music Broadcast trunk TNs is not checked against the ISM limit at SYSLOAD. The Music Broadcast ISM limit pertains to Music Broadcast connections only and not to TNs. Figure 66 is an example of the updated header:

Figure 66
ISM header in Overlay 14

TNS	AVAIL: xxxxx	USED: xxxxx	TOT: xxxxx
MUS CON	**AVAIL: xxxxx**	**USED: xxxxx**	**TOT: xxxxx**

Option 11C and Input-Output Disk Unit with CD-ROM (IODU/C) customers can modify ISM parameters via keycode. A keycode is a machine-generated digitally signed list of customer capabilities and authorized software release. A security keycode scheme protects ISM parameters.

In order for Option 11C and IODU/C customers to expand ISM limits, they must order and install a new keycode. This installation is performed using the Keycode Management feature. All Keycode Management commands are executed in Overlay 143. To make the expansion effective, the customer must sysload. For further information on keycode installation, please refer to *Software conversion procedures*.

For customers without Option 11C or IODU/C, ISM parameters are delivered as per existing operation.

For further information on ISM, refer to the Incremental Software Management feature module in *X11 features and services*.

Traffic Study Option

The Traffic Period Option (TPO) allows a customer to enhance their TFC002 reports to accumulate trunk usage data after every traffic period instead of accumulating usage only after a call disconnects. With this option enabled in Overlay 17, the Common Channel Signaling (CCS) associated with lengthy calls is reported in each traffic report interval throughout the duration of the call.

Previously, this feature did not apply to RAN and Music trunks. With the introduction of Music Broadcast, however, a Music call may last for an extended period of time. Therefore, changes are made to the Trunk Traffic Reporting Enhancement with the introduction of the TFC111 traffic report.

The TFC111 report provides information on the usage of broadcasting routes. For the TFC111 report to be output, customer report number **11** must be selected using the SOPC command in Overlay 2. For example, for Customer 0, SOPC 0 11 is entered. To print the TFC111 report, the TOPC command in Overlay 2 is used. For example, for Customer 0, TOPC 0 11 is entered. The TFC 111 report is also printed when automatic traffic reports are scheduled in Overlay 2.

The System Traffic message, TFS 0503, is output each time a music request cannot be completed because the total number of active Music Broadcast connections is equal to the system's ISM limit. Figure 67 is an example of the customer report, TFC111, for Music Broadcast routes.

Figure 67
TFC111 Report for a broadcasting music route

0200 (System ID)	TFC111	
000 (Customer number)		
030 (Route number)	MUS (Trunk type)	
001132 (Successful broadcast connections peg count)	00016 (Average call duration)	00000 (Unsuccessful broadcast connections peg count)
00000 (Broadcast connections peg count for lowest usage trunk)	00000 (Broadcast connections peg count for second lowest usage trunk)	00002 (Broadcast connections peg count for third lowest usage trunk)

Operating parameters

Music Broadcast requires any Music trunk and an external music source **or** a Meridian Integrated RAN (MIRAN) card (NTAG36). MIRAN has the capability to provide audio input for external music.

A Conference loop is not required for Music Broadcast.

With the Music Broadcast package configured, both existing Conference-based Music and Music Broadcast can co-exist on the same system. The type of Music is dependant upon the BDCT prompt in the Route Data Block.

The Music Broadcast feature is applicable to Music routes only.

To upgrade an existing non-broadcasting Music route to a broadcasting Music route, the REQ prompt must be set to CHG and the BDCT prompt must be set to YES in Overlay 16.

A broadcasting Music route may only be changed to a non-broadcasting Music route if it is first removed in Overlay 16 and then added back into the system as a non-broadcasting Music route by setting the BDCT prompt to NO. In this case, the CFLP prompt in Overlay 14 must be defined, and the Conference loop number for the non-broadcasting Music route must match the loop number that was set previously in Overlay 17.

If more than 64 Music Broadcast connections are required due to high traffic, additional trunks, each with up to 64 Music Broadcast connections, can be added. This same Music source can be cross-connected to all Music trunk TNs within a particular Music Route.

When more than one Music trunk is attached to a broadcasting Music route, a trunk is first sought within the caller's own group. An already active trunk is chosen initially in order to give music to the requesting party. If there is not a Music trunk that is already active or if all active Music trunks already have the maximum number of callers connected, an idle trunk is sought. If an idle trunk is found, the call is connected. If there are no trunks available within the caller's group, trunks in other groups are sought.

Although Music Broadcast supports inter-group music, it is advisable that for multi-group systems with high inter-group traffic, a Music trunk be provisioned in each network group to reduce junctor traffic.

Several routes can be supported via Music Broadcast; hence, different types of Music are also supported. On multi-group systems, however, network group junctor traffic limitations may cause difficulty in supporting several types of music on one system. In this case, additional trunks and additional connections can be added to the system.

If blocking occurs, silence or ringback tone is given by the features requesting music.

When the actual number of Music Broadcast connections in use is equal to the ISM limit, another connection is not allowed. In this case, the Blocking operation is retained. Therefore, silence or ringback tone is given by the features requesting music. This information is output in the Traffic report (TFS 0503).

The total number of Music Broadcast trunks multiplied by the maximum number of Music Broadcast connections per trunk may be greater than the ISM limit. The ISM limit of Music Broadcast connections is shared between different types of Music routes.

When a Music Broadcast trunk port is forced to disconnect through maintenance, all connected callers hear silence but remain on hold.

Only those calls receiving music in an ACD queue are restored by the INIT ACD Queue Call Restore feature following a system initialization. Any other calls receiving music are dropped, and the callers hear silence.

Feature interactions

Call Detail Recording

Due to the number of callers that can be connected to a broadcasting Music trunk at one time, Call Detail Recording (CDR) is not supported on broadcasting Music routes.

> ***Note:*** CDR is prompted for Music routes in Overlay 16. However, even if CDR is set to YES, a CDR record will not be output for Broadcasting Music routes.

Integrated Call Center Management

The Integrated Call Center Management (ICCM) broadcast capability on a Meridian 1 is independent of the Music Broadcast capability which is applicable only to Music routes. This ICCM broadcast capability applies only to Interactive Voice Response (IVR) voice ports.

The script command GIVE MUSIC <music route number> connects a call to the specified Music route. The Music Broadcast feature is applied if appropriate.

The script command GIVE BROADCAST ANNOUNCEMENT {NOT INTERRUPTIBLE} <acd_dn> {WITH TREATMENT <treatment>} applies to IVR ports only, and the ICCM broadcast capability is applied in this case.

Meridian Interactive Voice Response

Interactive Voice Response (IVR) interacts with the Music Broadcast feature, using the existing functionality of a non-broadcasting Music Route.

Recorded Announcement Broadcast

The Recorded Announcement (RAN) Broadcast feature is applicable to RAN only, and the Music Broadcast feature is applicable to Music only.

Feature packaging

Music Broadcast (MUSBRD) is package 328. The following packages are also required to provide Music Broadcast capability:

— Music (MUS) package 44

— Recorded Announcement (RAN) package 7

To provide Music Broadcast capability to Enhanced Music (EMUS) features, the Enhanced Music (EMUS) package 119 is also required.

Feature implementation

Music Broadcast

LD 16 – Upgrade an existing non-broadcasting Music route to a broadcasting Music route.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
ROUT	0-511 0-127	Route Number. For Option 11C.
TKTP	MUS	Music Trunk Data Block.
ICOG	OGT	Outgoing only Trunk.
...		
BDCT	YES	Allow Broadcast capability. NO = Deny Broadcast capability (default). If BDCT = YES, no conference loop is required. Each Music trunk has 64 broadcast connections.

Conference-based Music

LD 16 – Define a Conference-based Music route.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
ROUT	0-511 0-127	Route Number. For Option 11C.
TKTP	MUS	Music Trunk Data Block.
ICOG	OGT	Outgoing only Trunk.

...		
BDCT	NO	Deny Broadcast capability. YES = Allow Broadcast capability. If BDCT = YES, no conference loop is required. Each Music trunk has 64 broadcast connections.

LD 14 – Configure new Conference-based Music trunks.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	MUS	Music trunk.
TN	l s c u c u	Terminal number. For Option 11C.
...		
RTMB	0-511 1-510 0-127 1-510	Route number and Member number. For Option 11C. The Route number is specified in LD 16.
...		
CFLP	0 - 158	Music Conference Loop. Prompted only for non-broadcasting Music routes.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	12
Networking:	No

Music, Enhanced

Enhanced Music (EMUS) provides music for internal and external calls. Music is provided when telephones are placed on Hold, Consultation Hold, and Camp-On and when calls at the Attendant Console are split using the "Exclude Source/Destination" keys.

Enhanced Music (EMUS) provides music in situations described in Table 115.

Table 115
Features vs. no Music, Music, and Enhanced Music

	Without Music		Music		Enhanced Music	
	Sets	Trunks	Sets	Trunks	Sets	Trunks
ROA Waiting	No	No	Yes	Yes	Yes	Yes
Call Park	No	No	Yes	Yes	Yes	Yes
ACD Music	No	No	Yes	Yes	Yes	Yes
Hold Key	No	No	No	Yes	Yes	Yes
Permanent Hold	No	No	No	Yes	Yes	Yes
Consultation Hold	No	No	No	No	Yes	Yes
Splitting	No	No	No	Yes	Yes	Yes
Camp-On	No	No	N/A	Yes	N/A	Yes

Operating parameters

The requirements for Enhanced Music on Hold are the same as for Music on Hold. See "Music" on page 2045.

Trunks receive Music on a route basis. Telephones receive Music on a customer basis.

Feature interactions

Enhanced Music on Hold has the same feature interactions as Music on Hold. In addition, it has interactions with the following features:

Attendant Busy Verify

When the attendant attempts to Busy Verify a telephone receiving Music, the Music is removed. When the attendant releases, Music is returned.

Call Hold, Deluxe

A caller placed on Hold by a member of a multiple appearance group receives Music regardless of whether the call is on Hold or Exclusive Hold.

Call Transfer

The held party receives Music when the other party presses the Call Transfer key. The Music connection remains until the Call Transfer key or the DN key is pressed, ending the Consultation Hold state.

Charge Account and Calling Party Number

The Charge Account (CHG) and Calling Party Number (CPN) keys place the far end party on Hold while a charge number is entered. The held party receives Music during this period.

Conference

The held party receives Music when the Conference key is pressed, while the conference is being established, and whenever the conference is reduced to two parties with one party on Hold. Once the conference is established, Music is no longer provided.

A Six-party Conference operates the same as a Three-party Conference.

Privacy Release

When using Privacy Release to add one or more members to a call already receiving Music, the Music is removed.

Telephones - M3000

The Switch Parties key allows Music to the party on Hold and ends Music to the other party each time it is pressed.

Feature packaging

Enhanced Music (EMUS) package 119 requires:

— Music (MUS) package 44, and

— Recorded Announcement (RAN) package 7.

Feature implementation

LD 17 – Add or change Conference loops for Music on Hold.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN CEQU	Configuration Record. Release 19 gate opener.
CEQU	(NO) YES	Change to CE parameters.
- XCT	0-158	Loop number for NT8D17 Conference/TDS/MFS card. Enter an even network loop number for TDS/MFS functions. The conference function is automatically assigned the next higher (odd) loop number.
- CONF	0-158	Loop number for conference card (must be an even numbered loop).

LD 15 – Enable/disable Music for a customer.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
CUST	0-99 0-31	Customer number. For Option 11C.
- MUS	(NO) YES	Enhanced music for telephones.
- MUSR	0-511	Music route for telephones.

LD 16 – Add or change a Music route.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RDB	Route data block.
CUST	0-99 0-31	Customer number. For Option 11C.
TKTP	MUS	Music route.
ICOG	OGT	Outgoing route only.
ACOD	xxxx	Trunk route access code.
Note: All other prompts can be set to default values.		

LD 14 – Add or change a Music trunk. At least one Music trunk per network group is required for each customer requiring Music.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	MUS	Music trunk.
TN	l s c u c u	Terminal Number. For Option 11C.
RTMB	xxx yyy	Route and member number.
CFLP	0-158	Conference loop assigned to music in LD 17.

LD 16 – Enable/disable Music on Hold for trunk routes.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RDB	Route data block.
CUST	0-99 0-31	Customer number. For Option 11C.
TKTP	COT DID FEX TIE WAT	Trunk type.
MUS	(NO) YES	Music on Hold (is not) is to be provided for this trunk route.
MRT	0-511	Music route number.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	14.46E
Networking:	No

M2312 Digit Display

This feature supports the addition of a two line by 24-character digit display to the M2112 telephone, making it possible to assign an M2112 digital telephone one of the Display Allowed (ADD or DDS) Classes of Service. With either of these classes assigned, the M2112 digital telephone with digit display (M2312) will display digits in a manner similar to the SL-1 telephone.

The display format is the same as that used for the SL-1 telephone, except that 24 characters are available (instead of 16).

Operating parameters

The M2312 telephones will operate on either double density or quadruple density loops. Those telephones configured on a double density loop will be capable of voice service only. Those telephones that are configured on quadruple density loops can provide integrated voice and data services.

Feature interactions

Call Party Name Display

The calling party number can be displayed only when the call is active.

Hold

The digit display will go blank when a call is placed on hold.

Mute

Muting a call will not affect the digit display.

SL-1 digit display

The first line of the digit display can show the characters 0-9, *, #, P, and - The M2312 digit display differs in this respect from that of the SL-1 telephone; the M2312 can display * and #, and hence does not use a space or the H character to represent them.

Feature packaging

M2000 Digital Sets (DSET) package 88.

Feature implementation

LD 11 – Create or modify the data blocks for Meridian 1 proprietary telephones.

Prompt	Response	Comment
TYPE	xxxx	Digital data block for xxxx digital set.
CLS		Class of Service.
	(NDD) ADD DDS	No Digit Display, Automatic Digit Display, or Digit Display Standard.

Feature operation

The first line will be capable of displaying the same characters as the SL-1 telephone's digit display. The second line will display the date and time. In addition, when a call is active on key 0, a call timer will be displayed on the second line.

The following display lines can be called up by manual key operations:

— date and time

— buzz DN

— call waiting party

— voice call party

— autodial number

— speed call number

— ring again party

— call forward party

— call pickup

The following display lines can be automatically displayed:

— dialed number, and

— number of calling party.

The time and date function shown on the second display line is generated within the telephone. However, the telephone clock is automatically updated at least once a day from the switch's system clock. The call timer that appears on the second line is generated and controlled completely within the telephone. The function is not controlled by the switch.

N Digit DNIS

Dialed Number Identification Services (DNIS) presents the Automatic Call Distribution (ACD) call to an agent's set or terminal. The incoming call displays the DNIS digits which represent product lines or services. The displayed DNIS digits reduces the time needed to service a call and the additional information helps the agent provide a greater degree of customer service. The ACD Routing by DNIS number routes the call to a specific ACD DN based on the DNIS number dialed.

With the N Digit DNIS feature, the DNIS length expands to a maximum number of 31 digits. Both ACD and Network ACD (NACD) support the N Digit DNIS feature.

Operating parameters

If the system initializes during an active call, DNIS information is lost.

M911 trunks cannot be configured as DNIS trunks.

X11 system messages for the Time Slot Monitor (TSM) supports 31 digits of DNIS.

To use the full functionality of N Digit DNIS with NACD, all systems must be running X11 Release 24 software. Other scenarios are described in the following:

— If an NACD system is running X11 Release 24 software or later on one node and pre X11 Release 24 on another node, the DNIS information display is one to seven digits.

— If a system node is running pre X11 Release 20 software, the DNIS routes are configured for three to four digits.

Applications and features display DNIS in the following ways:

- Meridian MAX 9.0 and later supports up to nine digits of DNIS information. Nine digits of DNIS information are sent over the High Speed Link (HSL). The first or last nine digits of DNIS information is sent depending on the configuration of the WDGT prompt in the RDB block.
- Auxiliary Processor Link (APL) supports four DNIS digits. If the DNIS information is longer than four digits, the first or last four digits are sent over the APL depending on the configuration of the WDGT prompt in the RDB block.
- Call Detail Recording (CDR) supports up to seven digits of DNIS digits. If more than seven digits of DNIS are received, the first or last seven digits are displayed on the CDR, depending on the configuration of the WDGT prompt in the RDB block.
- The CDR format on pre X11 Release 18 software (or NCDR = NO) supports the display of four digits. First or last seven digits are displayed, depending on the configuration of the WDGT prompt in the RDB block.
- Call Party Name Display (CPND) supports name configuration up to seven digits of DNIS. If the DNIS information is more than seven digits, a name is not configured.
- Feature Group D supports seven digits of DNIS information.
- The agent's set is limited to 12 digits of DNIS display. If more than 12 digits of DNIS are received, the first 12 or the last 12 digits of DNIS are displayed, depending on the configuration of the WDGT prompt in the RDB.

Feature interactions

Automatic Call Distribution DNIS routing through IDC table

The Incoming Digit Conversion (IDC) table converts the DNIS digits to a valid DN. With the N Digit DNIS feature, the DNIS information is expanded to a range of one to31 digits. The maximum number of DNIS digits that are translated by the IDC tree to an internal DN is limited to 16, due to the DC feature.

Application Module Base

Meridian 1 is connected to Application Module Base (AM Base) through Application Module Link (AML). DNIS information is in AML messages; therefore, the AM Base supports the expanded DNIS information.

Application Module Link (AML) messages

Call presentation and call modification receives DNIS through AML messages. Messages related to DNIS go through the AML to the Meridian Link Module to the Customer Controlled Routing (CCR).

Call Detail Recording

The Call Detail Recording supports up to seven DNIS digits. If the DNIS digits exceeds seven digits, the Call Detail Recorder (CDR) uses the first or last seven digits, depending on the configuration of the WDGT prompt in the RDB block.

Customer Controlled Routing

Customer controlled routing (CCR) uses the DNIS number to determine which call processing treatment is used for a DNIS trunk call.

Digit display for DNIS

The agent set is limited to a display of 12 DNIS digits. If the digits exceed the set's display capabilities, the first or last 12 DNIS digits are displayed depending on the configuration of the WDGT prompt in the RDB block.

Host Enhanced Routing

The Meridian Link's Host Enhanced Routing allows an incoming call to be routed before call termination. An Incoming Call (ICC) message sent to the Meridian Link Module contains calling party information, DNIS information, and Controlled Directory Number (CDN).

Meridian Link Interactions

Any ringing message sent to the Meridian Link over the AML contains expanded DNIS information. The Meridian Link sends this expanded information to the host application.

Meridian Mail

Meridian Mail receives DNIS digits over the Command Status Link (CSL). The DNIS message contains one to 31 DNIS digits, instead of the previously supported one to seven digits. Since Meridian Mail limits DNIS digits to 30, the AML message uses 30 digits.

Meridian MAX

Meridian 1 communicates with Meridian MAX, ACD MAX, or ACD AUXILIARY through High Speed Link (HSL). Meridian MAX 8.0 or later supports nine digits of DNIS.

Multi-Frequency Signaling for KD3 for Spain.

If a DNIS route uses Multi-Frequency Compelled (MFC) signals, the DNIS route must use the same number of digits as the MFC.

Multi-Frequency Signaling for Socotel

Multi-Frequency signaling for Socotel (MFE) trunks use either four or five signals, this requires DNIS to use the same numbe

Network Automatic Call Distribution

The Network Automatic Call Distribution (NACD) sends and receives DNIS calls to a remote node through an NACD-Call Setup message. The remote node receives and saves the expanded one to 31digits of a DNIS message.

Symposium Call Center Server

The interaction of N Digit DNIS with Symposium Call Center Server (SCCS) is the same as its interaction with Customer Controlled Routing (CCR) and AM Base. Any AML message sent to the AM Base contains expanded DNIS information. AM Base supports the expanded DNIS information. Symposium supports seven digits of N Digit DNIS information.

Feature packaging

This feature is packaged as part of the existing DNIS package 98.

Feature packages required for the N Digit DNIS are:

— Dialed Number Identification System (DNIS) package 98

— Automatic Call Distribution (ACD A) package 45

— Digit Display (DDSP) package 19

— Incoming DID Digit Conversion (IDC) package 113

— New Format Call Detail Recording (FCDR) package 234

— New Flexible Code Restriction (NFCR) package 49

Feature implementation

The Route Data Block is modified to allow up to 31 digits for a DNIS prompt.

Configure the following overlays for full functionality of DNIS messaging:

— Define SDI port for Auxiliary Processor Link in Load 17.

— Define Incoming Digit Conversion table in Load 49.

and

— Define IDC-DNIS route in Load 16.

OR

— Define a trunk that auto-terminates on ACD-DNIS in Load 14.

and

— Define a route Auto Terminate Route in Load 16.

— Define APL Link number, enable the Incoming Digit Conversion (IDC) operation to include DNIS for a customer in Load 15.

— Define ACD group in Load 23.

LD 17 – Define SDI port for Auxiliary Processor Link.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration Record.
ADAN	NEW TTY 0-15	Add an APL port.
CTYP	aaaa	Card type. aaaa = DCHI, SDI, SDI2, SDI4.
USER	APL	APL port connects to data link.

LD 49 - Define Incoming Digit Conversion (IDC) table.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	IDC	Type of data block (FCR or IDC).
CUST	xx	Customer number xx as defined in LD 15.
DCNO	0-254	Incoming Data Conversion (IDL) tree number.
...	...	...
...	...	...
IDGT	0-99999999 0-99999999	Incoming digits to be converted to ACD DN.
	<CR>	Re-prompt request.

AND

LD 16 - Define Incoming DID Digit Conversion DNIS route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route data block.
CUST	xx	Customer number xx as defined in Load 15.
ROUTE	 0-127 0-511	Route number. This range applies to Option 11C. This range applies for machines 51C, 61C, 81, 81C.
...	...	...
AUTO	 NO	Auto-terminate. YES = The route members terminate on DN defined by response to ATDN prompt in Load 14. NO = The route members terminate normally.
DNIS	YES	ACD DNIS route.
--NDGT	1-(4)-31	Number of DNIS digits required on the route. The extension 31 digits is available only for DID, TIE or IDA routes.
--WDGT	(L)F	First or last DNIS digits to be sent on APL and HSL link. Where: F = First, L = Last (default) WDGT has no effect on AML Links. All DNIS digits are sent for AML. Prompted if NDGT is greater than four. Also used for CDR when the New Format CDR (FCDR) package 234 is disabled. First or last 4 digits for APL. First or last 12 DNIS digits for digit display. First or last 9 DNIS digits for MAX. First or last 7 DNIS digits for CDR.
--IDC	YES (NO)	Incoming DED digit conversion on this route YES = Allow Incoming DID Digit Conversion on this route. (NO) = Deny Incoming DID Digit Conversion on this route.

--DCNO	0-254	IDC translation table for this route in the day mode.
--NDNO	0-254	IDC Conversion Table for the night mode.

OR

LD 14 – Define a trunk that auto-terminates on ACD-DNIS.

Prompt	Response	Description
REQ	NEW	Add a trunk.
TYPE	DID	Direct Inward Dialing trunk type.
RTMB	x y	x = Route number defined in LD 16. y = member number.
ATDN	xxxx	xxxx = ACD-DN defined in LD 23.
CLS	DTN	Digitone signaling.

AND

LD 16 – Define a route with DNIS feature enabled and AUTO-terminate.

Prompt	Response	Description
REQ	NEW	Add a new data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number xx as defined in LD 15.
ROUT	 0-127 0-511	Route number. 0-127 = This range applies to Option 11C. 0-511 = This range applies for machines 51, 51C, 61, 61C 71, 81, and 81C. ***Note:*** In X11 Release 14 and later, route 31 is no longer reserved as a Private route but can be configures as one.
...	...	

AUTO	YES	Auto-terminate trunk. YES =YES = the route members terminate on DN defined by response to Auto Terminate Directory Number prompt in D 14. (NO) =The route members terminate normally at the console.
DNIS	YES	ACD-DNIS route. YES = allow the ACD DNIS route. (NO) = Deny the ACD DNIS route. Prompted with Automatic Call Distribution Package D. (ACCDD) package 50, and the RTYP = TIE or Direct Inward Dialing (DID).
NDGT	1-(4)-7 1-(4)-31	Number of DNIS digits required on the route. The extension to 31 digits is available only for DID, TIE or IDA routes.
WDGT	(L) F	First or last 4 DNIS digits to be sent on APL and HSL link. WDGT has no effect on AML links. All DNIS digits are sent for AML.. Prompted if NDGTR is greater than 4. Also used for CDR when the New Format CDR (FCDR) package 234 is disabled. ***Note:*** The number of (MFX), MFE or MFC digits takes precedence over the number of DNIS digits that are configured.

LD 15 – Define APL Link number, enable the Incoming Digit Conversion (IDC) operation to include DNIS for a customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:		Disable/Enable New flexible code Restriction.
	FCR	Flexible Code restriction.
CUST:	xx	Customer number.
NFCR		New Flexible Code Restriction. (NO) = Default, disable New Flexible Code Restriction.
	YES	YES = Enable New Flexible Code Restriction. To build an Incoming Digit Conversion (IDC) table in LD 49, NFCR and Incoming DID Digit Conversion (IDCA) must be set to YES. NFCR is prompted with New Flexible Code Restriction (NFCD) package 49.
-MAXT	1-255	Maximum number of New Flexible Code Restriction (NFCR) tables. Once defined a lower value cannot be entered for MAXT. The sum of the values for MAXT + DCMX ≤ 255 per customer.
IDCA		Incoming DID Digit Conversion.
	(YES	(NO) = Default. Deny Incoming DID Digit Conversion. YES = Allow Incoming DID Digit Conversion. NFCR must = YES before IDCA can = YES. Prompted with Incoming Digit Conversion (IDL) package 113.
-DCMS	1-255	Digit conversion maximum number of tables (DCMS). The sum of the values for MAXT and DCMX cannot exceed 255 or MAXT + DCMX = 255.

LD 23 – Define ACD group.

Prompt	Response	Description
REQ	NEW	Add ACD group.
TYPE	ACD	ACD data block.
CUST	xx	Customer number.
ACDN	xxxx	ACD Directory Number.

Feature operation

No specific operating procedures are required to use this feature.

Network and Executive Distinctive Ringing

Network Distinctive Ringing (NDRG) allows a distinctive ringing cadence to be configured throughout a Meridian network. Distinctive ringing is defined on a route basis. There are four NDRG distinctive ringing cadence indexes that can be defined for a route. These indexes are contained in the Flexible Tone and Cadences (FTC) table. If one of these indices has been defined for a route and an incoming trunk call over that route terminates on the local node, the terminating set receives distinctive ringing. If the incoming call tandems to another node via an Integrated Services Digital Network (ISDN) TIE trunk, the terminating set at the terminating node receives distinctive ringing if the TIE trunk has been marked as distinctive and if the NDRG feature is equipped at the terminating node – otherwise, normal ringing is given.

Executive Distinctive Ringing applies to both network and standalone environments. This feature allows a Class of Service to be entered for a telephone set, marking the set as "executive". When a call is made from an executive set, the called set is rung distinctively. This feature uses the distinctive cadences introduced by the Network Distinctive Ringing (NDRG) feature.

One of five Classes of Service can be entered – EXR1, EXR2, EXR3, EXR4, or EXR0. EXR is the Class of Service mnemonic that marks the set as executive and the digits one to four indicate which of the four distinctive ringing cadences to be applied. EXR0 is the default and marks a set as normal.

Operating parameters

Both Network Distinctive Ringing and Executive Distinctive Ringing can be equipped for a set. In this case, a cadence that is selected for NDRG can also be selected for EDRG.

Within a network, if there are five routes marked as distinctive, and if an incoming call tandems between two nodes that are connected by a single TIE trunk, the terminating node can provide unique distinctive ringing for only four of the five routes. The originating node can provide unique distinctive ringing to all five routes since each route can use a different Flexible Tone and Cadence (FTC) table.

Feature interactions

Conference

If a new party is to be included in an established conference, the ringing that is applied to the set of the new party depends on the sets of the established parties. The system scans the trunks and sets of the conferees for a trunk marked as distinctive or a set designated as executive. The ringing cadence of the new set depends on the highest index found by the scan.

Incoming trunk

An incoming trunk call that is redirected or attendant-extended will ring distinctively at the terminating set, according to the cadence index of the originating trunk route. If the terminating set is located at another node, it will ring distinctively according to the cadence index of the originating trunk route, if the NDRG feature is equipped at the terminating node.

Manual Signaling (Buzz)

Network Distinctive Ringing and Executive Distinctive Ringing do not affect the buzzing of a set.

Feature packaging

Network and Executive Distinctive Ringing is not packaged; however, the following packages are required to make it operational:

— Executive Distinctive Ringing (EDRG) package 185

— Distinctive Ringing (DRNG) package 74

— Flexible Tones and Cadences (FTC) package 125

— Integrated Services Digital Network (ISDN) package 145

— Integrated Services Digital Network Supplementary Features (ISDNS) package 161

Feature implementation

LD 10 – Respond to the CLS prompt to define the distinctive ringing cadence/tone to be used for analog (500/2500 type) telephones.

Prompt	Response	Description
...		
CLS	(EXR0), EXR1, EXR2, EXR3, EXR4	Executive Distinctive Ringing off (0). The digit indicates which of the four distinctive ringing tones and cadences defined in LD 56 is to be used.

LD 11 – Respond to the CLS prompt to define the distinctive ringing cadence/tone to be used for Meridian 1 proprietary telephones.

Prompt	Response	Description
...		
CLS	(EXR0), EXR1, EXR2, EXR3, EXR4	Executive Distinctive Ringing off (0). The digit indicates which of the four distinctive ringing tones and cadences defined in LD 56 is to be used.

LD 16 – Deny or allow distinctive ringing.

Prompt	Response	Description
...		
DRNG	(NO), YES	(Deny) allow Distinctive Ringing for incoming calls. Distinctive Ringing only applies to CAM, COT, DID, FEX, TIE, and WAT trunks. These trunks cannot be configured as outgoing only in response to prompt ICOG.
NRDI	(0)-4	Network Distinctive Ringing Index. 0 = Default/undefined.

LD 56 – Define the distinctive ringing cadence for analog (500/2500 type) telephones and the distinctive ringing tone for Meridian 1 proprietary telephones in the Flexible Tone and Cadence (FTC) table.

Prompt	Response	Description
...		
- NDR1 PBX	0-255	Network Distinctive Ring 1 cadence for analog (500/2500 type) telephones.
- NDR1 BCS		Network Distinctive Ring 1 cadence for Meridian 1 proprietary telephones.
- - TDSH	i bb c tt	Tone definition for systems equipped with Tone and Digit cards, where: i = internal (0), or external (1) source bb = burst cc = cadence, and tt = frequency. Prompts with the response i bb c tt define the internal/external source, burst, cadence and frequency/level respectively. Enter the decimal equivalent (0-15) of the TDS Hex code. The first field is usually 0. If an external source is used, the entry is 1 and the fourth field is 0-7 for the specified channel.
- - XTON	0-255	XCT tone code.
- NDR2 PBX	0-255	Network Distinctive Ring 2 cadence for analog (500/2500 type) telephones.
- NDR2 BCS		Network Distinctive Ring 2 cadence for Meridian 1 proprietary telephones.
- - TDSH	i bb c tt	See above.
- - XTON	0-255	XCT tone code.
- NDR3 PBX	0-255	Network Distinctive Ring 3 cadence for analog (500/2500 type) telephones.
- NDR3 BCS		Network Distinctive Ring 3 cadence for Meridian 1 proprietary telephones.

- - TDSH	i bb c tt	See above.
- - XTON	0-255	XCT tone code.
- NDR4 PBX	0-255	Network Distinctive Ring 4 cadence for analog (500/2500 type) telephones.
- NDR4 BCS		Network Distinctive Ring 4 cadence for Meridian 1 proprietary telephones.
- - TDSH	i bb c tt	See above.
- - XTON	0-255	XCT tone code.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	21
Networking:	Yes

Network Intercom (Hot Type D and Hot Type I Enhancements)

Hot Line enables a designated telephone to place calls to a predetermined destination that may be internal or external to the Meridian 1. The call does not require attendant intervention. When the handset is lifted, or when a preprogrammed key is activated, the system speed calls a preprogrammed DN. Hot Lines access a set of Terminal Numbers programmed by direct entry using LD 11, or by list entry such as System Speed Call (SCC) using LD 18. A Hot Line call that enters the ringing state, is the same as a normal call.

Prior to Release 21, there were two types of Hot Line keys (DN-based Hot Type D, and Speed Call List-based Hot Type L). Release 21 introduces another type of Hot Line key – Hot Type I, while also providing improvements to the existing Hot Type D. These two improvements function in both standalone and network environments.

Hot Type D provides the ability for Meridian 1 proprietary telephones to have two-way calls on specially designated keys (not on the DN keys) with other Meridian 1 proprietary telephones connected to Meridian 1 PBXs across a Meridian Customer Defined Network (MCDN) Integrated Services Digital Network (ISDN). A Hot Type D call can now terminate in three different modes: Voice, Ringing, and Non-ringing. With the Voice mode, speech path is automatically connected after a short ring. With Ringing and Non-ringing modes, the call must be manually answered by the called party; the difference between the two modes is that for Non-ringing no audible tone is given (the Hot Line key flashes to indicate the call).

Hot Type I

An option is available with Hot Type I to provide a No Answer Indication informing the called party that a Hot Line call was made during the called party's absence. If a Hot Line call cannot be completed on the Hot Line key, the calling party is informed via the set's display, and the call is completed over the network as a normally dialed call that attempts to terminate on the destination Prime DN.

Hot Type D Enhancement

Hot Type D allows more than one set to have the same target DN defined for the Hot Type D key. This enhancement includes Voice, Ringing, and Non-ringing termination modes, and the capability to leave a No Answer Indication in some situations (that is, the Hot Line key winks). A call terminating on an enhanced Hot Type D key operates the same as if terminating on a Hot Type I key, if the originating DN is the same as the target DN defined for the key. If it is not the same, but another Hot Line key exists on the set that has a target DN that matches the originating DN, a No Answer Indication is left on that key.

Operating parameters

Hot Type I calls are not allowed on analog (500/2500 type) telephones.

Hot Line keys cannot be defined for M3000 sets.

A Hot Line key must not be defined on a station without a Prime DN, and must not be defined on the primary key. If this is not done, the improved functionality will not work and the call is treated as a non-Hot Line call.

The network DN for Hot Type I and the No Answer Indication component of Hot Type D must be either a Coordinated Dialing Plan or a Universal Dialing Plan number that must terminate on a Prime DN of a Meridian 1 proprietary telephone; otherwise, the call is competed as a non-Hot Line call.

The network-wide application of Hot Type I is only applicable to nodes in a Primary Rate Interface (PRI), ISDN Signaling Link (ISL), Virtual Network Services (VNS), and Basic Rate Interface network. The originating and terminating nodes of the Network Intercom (Hot Type D and Hot Type I Enhancements) feature must be Meridian 1 switches running a minimum of Release 21 software. Tandem nodes in the private network must be Meridian 1 (minimum Release 21 software) or another MCDN node able to transfer Remote Operations Service Element (ROSE) facilities.

If the termination mode is voice and the called party is idle and the handsfree voice call (HZA in LD 15) is active, there is no indication to the software that the called party really answered the call. If any other key is pressed, the No Answer Indication is not left.

Hot Line keys must be defined with the same dialing plan.

The Hot Type D enhancements can also function across R2MFC or Digital Private Network Signaling System #1 (DPNSS1) trunks provided that the calling party's DN is provided.

Feature interactions

Attendant Blocking of DN

A Hot Type I key cannot be blocked by the attendant, because it has no DN.

Pressing a Hot Type D key that is attendant blocked establishes the call on the source side of the attendant.

Auto Hold

If a user who originated a Hot Type I call receives or makes another call on another DN, pressing that DN puts the established call on hold. If a user presses the Hot Type I key while a call is established on it, the call is placed on hold. If the Hot Type I key is pressed while a call is established on another DN, the established call is put on hold. If a station with automatic hold allowed Class of Service receives an Hot Line call, the user of that station can put the active call on hold by pressing the Hot Type I key or by making or answering another call on another key.

Automatic Call Distribution

Hot Type I calls cannot terminate on an Automatic Call Distribution (ACD) DN. A call attempting to terminate on an ACD DN receives overflow tone. Hot Line calls involving ACD sets must use the Hot Type D option.

Call Forward All Calls

Hot Type I calls respect or override all kinds of Call Forward features (Busy, No Answer, All Calls, Internal, etc.) according to per-set definitions. If Call Forward is respected, the call becomes a normally dialed call and the originator will receive the appropriate indication on their display.

Call Forward and Busy Status

In a Secretarial filtering scenario, the secretary's Busy/Forward Status (BFS) lamp also will reflect that the boss' set is busy if the boss is on a Hot Type I call.

Call Join

Hot Type I calls can be moved to the Conference key with the Call Join feature.

Call Park

Hot Type I calls cannot be parked.

Call Party Name Display

Hot Type I calls display names the same as a normal call.

Hot Type I calls that become a normal call indicate on the originating station's display that the call is no longer a Hot Line call.

Call Pickup

Hot Type I calls cannot be picked up. An attempt to pick up a Hot Type I call results in an overflow tone.

Call Transfer

Hot Type I calls may be transferred to another Hot Line key or to a normal DN key; likewise calls on a normal DN key may be transferred to a Hot Line key.

Display Key

Hot Type I calls are supported by the Display key feature; pressing the Display key and then the Hot Type I key will show the target DN on the originating station's display.

Do Not Disturb

Hot Type I calls ignore the Do Not Disturb (DND) feature. Hot Line calls are presented to the defined target, even when DND is activated.

Flexible Feature Code (FFC) Boss Secretarial Filtering

Hot Type I calls overrides this feature (i.e., Hot Type I calls are not filtered by FFC Boss Secretarial filtering). The call terminates on the Boss' set and is not forwarded to the secretary.

FFC Boss Secretarial Filtering takes precedence over enhanced Hot Type D calls. In this case, if FFC Boss Secretarial Filtering is active, calls terminate on the secretary's set.

Last Number Redial

A Hot Line key cannot be redialed using the Last Number Redial feature.

Make Set Busy

Hot Type I calls terminating on a station in the Make Set Busy mode override Make Set Busy.

Multiple Appearance Redirection Prime

If more than one set is allocated the same prime DN, the Hot Type I call will terminate on the set designated as the Multiple Appearance Redirection Prime (MARP). If the MARP DN is not the prime DN on the set, or if the set designated as the MARP DN is not a Meridian 1 proprietary telephone, the first Meridian 1 proprietary telephone with the prime DN will be used. If none of these conditions are met, the call will terminate as a non-Hot Line call and the calling party will be notified on the display.

Hot Type D calls can have voice termination only on a MARP Terminal Number (TN), or if there is no MARP TN, then on the first TN in the TN list. A No Answer Indication for Hot Type D can only be left on the MARP TN, or if there is no MARP TN, then on the first TN in the TN list.

Override

An internal Hot Type I call never returns busy, unless the call became a non-Hot Line call due to the Hot Line key being busy. In this case, the call behaves like a normally dialed call, and override can be used upon receipt of a busy signal.

Ring Again

Hot Line calls terminating on a busy key become normal calls. Hence, they may use the Ring Again feature under normal circumstances.

Ring Again on No Answer

If Ring Again on No Answer is activated for a Hot Type I call, it is activated as if the call had been dialed normally.

Ringing Change Key

The ringing/non-ringing mode of an enhanced Hot Type D or of a Hot Type I key is not changeable by using the Ringing Change Key feature.

Secretarial Filtering

In a Secretarial filtering scenario, the secretary's BFS lamp also will reflect that the boss' set is busy if the boss is on a Hot Type I call.

Vacant Number Routing

Hot Types I keys and enhanced Hot Type D keys support vacant number routing.

Feature packaging

The Network Intercom (Hot Type D and Hot Type I Enhancements) feature is included in Enhanced Hot Line (HOT) package 70.

For Hot Type I in an ISDN network the following packages are required:

— Advanced ISDN Network Services (NTWK) package 148

— Integrated Services Digital Network (ISDN) package 145

— and at least one of the following 1.5 Mbps Primary Rate Interface (PRA) package 146, Integrated Services Digital Network Signaling Link (ISL) package 147, 2.0 Mbps Primary Rate Interface (PRI2) package 154, Virtual Network Services (VNS) package 183, and ISDN BRI Trunk Access (BRIT) package 233.

DNPSSS1 connectivity for Hot Type D requires:

— Integrated Digital Access (IDA) package 122

— Digital Private Signaling System 1 (DPNSS) package 123

R2MFC connectivity for Hot Type D requires:

— Multifrequency Compelled Signaling (MFC) package 128

Feature implementation

LD 11 – Define Hot Type D and I keys and Classes of Service as follows:

Prompt	Response	Description
REQ	NEW, CHG, OUT	Configure, change or remove.
TYPE	aaaa	Telephone type where aaaa cannot = M3000.
TN	l s c u c u	Terminal Number. Terminal Number for the Option 11.
...		
CLS	FICA FICD NAIA NAID	Forward Hot Type I allowed. Forward Hot Type I denied. No Answer indication allowed. No Answer indication denied.
...		
KEY	nn HOT D dd target_num hot_dn R/V/N/(H)	Two-way Hot Line Key. dd = number of digits dialed. target_number = terminating DN (31 digits maximum). Hot_dn = two-way Hot Line DN. Termination mode: R = Ringing V = Voice N = Non-ringing, and H = Hot Line (the default).

KEY	nn HOT I dd target_number (V)/N/R	Hot Line key. dd = number of digits dialed. target_number = terminating DN (31 digits maximum). Termination mode: V = Voice (the default) N = Non-ringing, and R = Ringing.

LD 95 – Configure the Calling Party Name Display:

Prompt	Response	Description
REQ	NEW, CHG	Configure or change.
TYPE	CPND	Calling Party Name Display.
0-99 0-31	Customer number. For Option 11C.	0-99 0-31
...		
NITC	aaaa (NI)	Non-Hot Line call. Indicates that the Hot Line call terminated as a normal call.

Feature operation

Press the Hot Type I or D key to initiate an Hot Line call to a target DN (the DN may be an external DN in an MCDN ISDN network).

The called party answers the call by pressing their Hot Type I or D key if configured. If the called party has no Hot Type I or D key configured, the call will behave as a normal call and is answered accordingly.

If the called party does not answer, and has No Answer Indication Allowed Class of Service, the Hot Type I or Hot Type D key will be winking as a form of No Answer Indication.

Introduced in X11 Release:	9.30A
Networking:	No

New Flexible Code Restriction

New Flexible Code Restriction (NFCR) controls the access of Toll Denied terminals to outgoing trunk routes and digits dialed on them. Calls are allowed or denied based on the specific digit sequence dialed.

Toll Denied (TLD, CTD, CUN) telephones and trunks are assigned a Network Class of Service (NCOS) and are allowed or denied calling privileges according to the Facility Restriction Level (FRL) assigned to their NCOS. If, however, a user who has CTD or CUN Class of Service has dialed the call using a Basic Alternate Route Selection (BARS), Network Alternate Route Selection (NARS), Coordinated Dialing Plan (CDP), or Automatic Number Identification (ANI) access code, the NFCR restrictions do not apply. For these users, NFCR applies only on direct trunk access code type calls. TLD users are always affected no matter how their call is dialed.

When a user accesses an outgoing route, the user's assigned FRL determines which digits are allowed or denied on that route. Up to eight FRL codes can be assigned per trunk route. When a user dials denied digits following direct trunk access codes, intercept treatment is given. NFCR can be programmed to deny certain outpulsed digits, not dialed digits, when Electronic Switched Network (ESN) calls are to be denied for TLD users.

Using "code restriction trees," NFCR can be programmed to analyze each digit individually and allow or deny a call on the basis of any digit or digit sequence dialed. There can be up to 255 code restriction trees per customer group. Each trunk route can access up to eight trees, and each tree can be used by more than one route. The code restriction tree corresponding to the terminal user's FRL is defined by the trunk route. Digits can also be bypassed and allowed to process with no restriction; however, certain digits that follow these might be restricted.

NFCR can be programmed to count the number of digits dialed and deny any call exceeding the specified number of digits. If a user dials an octothorpe (#) before NFCR has finished digit counting, the call is disallowed and intercept treatment is given. This prevents digits from 2500 sets or Dual-tone Multifrequency (DTMF) trunks from being outpulsed before being counted or analyzed by code restriction. Up to 50 digits can be analyzed.

Operating parameters

New Flexible Code Restriction (NFCR) can be programmed to count the number of digits dialed and deny any call exceeding the specified number of digits.

Only the digits zero (0) through nine (9) are considered. If a user dials an asterisk (*), it is not counted as a dialed digit. If the user dials an octothorpe (#) before NFCR has finished digit counting, the call is disallowed and the appropriate intercept treatment is provided. This prevents digits from 2500-type telephones or Dual-tone Multifrequency (DTMF) trunks from being outpulsed before being counted or analyzed by code restriction.

As many as 255 code restriction trees are available per customer. Eight code restriction trees can be referenced by each trunk route.

Up to 50 digits can be analyzed by NFCR.

When Code Restriction (LD 19) and NFCR (LD 49) are both enabled for the same customer, NFCR takes precedence. Any parameters required for Code Restriction are ignored.

Feature interactions

Access Restrictions

The Code Restriction feature and New Flexible Code Restriction cannot be implemented simultaneously for the same customer.

Attendant Blocking of Directory Number

When the attendant has a blocked DN on the source side and dials on the destination side, any New Flexible Code Restriction active for the set of the blocked DN will be overridden. This is the same as if the attendant had a normal established call to the DN on the source side and dials the destination side.

Authorization Code Security Enhancement

If the Class of Service of the authorization code is Toll Denied (TLD), NFCR is applied. If the Class of Service is Conditionally Unrestricted (CUN) or Conditionally Toll Denied (CTD) and the call is not routed through BARS/NARS, CDP or ANI, NFCR is applied.

Automatic Number Identification

Calls from Toll Denied (TLD) stations routed by Automatic Number Identification (ANI) are subject to NFCR. Calls placed by Conditionally Toll Denied (CTD) and Conditionally Unrestricted (CUN) Class of Service stations subject to ANI are treated as unrestricted calls.

Automatic Redial

Automatic Redial (ARDL) calls must pass New Flexible Code Restriction (NFCR) checks. If the redialed number is restricted, the ARDL request is cancelled.

Basic Alternate Route Selection (BARS)
Network Alternate Route Selection (NARS)
Coordinated Dialing Plan (CDP)

Only TLD telephones are subject to NFCR when calls are routed by BARS/NARS/CDP. CTD and CUN calls routed by BARS/NARS/CDP are not subject to NFCR treatment.

China – Flexible Feature Codes - Outgoing Call Barring

Outgoing Call Barring uses NFCR trees to define the digit sequences that are not allowed for each level of barring.However, OCB analyses all dialed digits, whereas NFCR only analyses digits outpulsed on trunks. This means that the same tree will not normally be usable for both features, unless only Coordinated Dialing Plan trunk calls are to be blocked for both features and no digit manipulation is done.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

Toll-denied users (CLS = TLD) may be subject to NFCR if they make a NARS call across the DPNSS1 UDP network. The New Flexible Code Restriction feature is supported in a DPNSS1 UDP network.

Direct Inward System Access

If the Direct Inwart System Access (DISA) DN has a TLD, CUN, or CTD Class of Service, calls made through DISA are eligible for NFCR treatment.

Electronic Lock Network Wide/Electronic Lock on Private Lines

With NFCR, toll denied stations are allowed or denied calling privileges according to the Facility Restriction Level (FRL) assigned to the NCOS defined in the protected line block. For a locked set, NCFR uses the FRL assigned to the CNCS to determine its calling privileges if one is defined; if no CNCS is defined, the NCOS of the locked set will be used.

Federal Communications Commission Compliance for Equal Access

The New Flexible Code Restriction (NFCR) feature has been modified to allow for the restriction of Equal Access international toll calls (10XXX+011+CC+NN) while not restricting Equal Access operator calls (10XXX+0).

Forced Charge Account

Calls placed through the Forced Charge Account feature are not eligible for NFCR treatment.

Network Class of Service

Toll Denied stations and trunks must have a Network Class of Service (NCOS) assigned to be allowed or denied calling privileges by NFCR. This is because the FRL associated with the NCOS of the user determines which codes are allowed or denied on an outgoing trunk call. The range of NCOS groups varies as follows:

(0)-3 for standalone CDP
(0)-7 for BARS/CDP and NFCR
(0)-15 for NARS and NFCR
(0)-99 for BARS/NARS/CDP/NFCR in X11 Release 13 and later software.

Scheduled Access Restrictions

Associating an FRL with a different NFCR tree affects any Network Class of Service (NCOS) that uses that FRL. Each such NCOS assigned to a Scheduled Access Restrictions (SAR) group might need to be reconsidered. Also, different facility restriction levels and NFCR trees are used at different times according to the NCOS assigned to the SAR group.

Feature packaging

New Flexible Code Restriction (NFCR) package 49 requires:

— Network Class of Service (NCOS) package 32.

Feature implementation

LD 15 – Enable NFCR for a customer.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB FCR	Customer Data Block. Release 21 gate opener.
CUST	0-99 0-31	Customer number. For Option 11C.
NFCR	(NO) YES	(Disable) enable NFCR.
- MAXT	1-255	Maximum number of code restriction trees.

LD 87 – Define NCOS groups and associated FRL.

Prompt	Response	Description
REQ	NEW CHG	Create new, or change.
CUST	0-99 0-31	Customer number. For Option 11C.
FEAT	NCTL	Network Control.
NCOS	(0)-99	NCOS group.

FRL	0-7	FRL is assigned to each NCOS. It determines the entries in a route list (RLI) to which it has access. 0 is the most restrictive, 7 is the least restrictive and can access more entries.

LD 49 – Add, change, or print code restriction trees.

Prompt	Response	Description
REQ	NEW CHG PRT	Create new, change, or print data.
TYPE	FCR	NFCR data block.
CUST	0-99 0-31	Customer number. For Option 11C.
CRNO	(0)-254	Code restriction tree number.
INIT	ALOW, DENY	Allow or deny all codes.
If INIT = ALOW the following prompts appear:		
DENY	xx...xx	Digit sequence to be denied.
ALOW	xx...xx	Digit sequence to be allowed.
BYPS	xx...xx	Digit sequence to be bypassed.
If INIT = DENY the following prompts appear:		
ALOW	xx...xx	Digit sequence to be allowed.
DENY	xx...xx	Digit sequence to be denied.
BYPS	xx...xx	Digit sequence to be bypassed

LD 16 – Associate an FRL with a code restriction tree.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
0-99 0-31	Customer number. For Option 11C.	0-99 0-31
ROUT	0-511	Route number.
FRL	x yyy	x = FRL number (0-7). yyy = code restriction tree number (1-255). FRL is reprompted to allow input of eight FRLs. A carriage return causes the next prompt to appear.

LD 10 – Assign an analog (500/2500 type) telephone a Toll Denied and Network Class of Service.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Optiuon 11C.
NCOS	(0)-99	NCOS.
CLS	TLD	Toll Denied Class of Service.

LD 11 – Assign Meridian 1 proprietary telephones a Toll Denied and Network Class of Service.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Optiuon 11C.
NCOS	(0)-99	NCOS.
CLS	TLD	Toll Denied Class of Service.

LD 14 – Assign a trunk a Toll Denied and Network Class of Service.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaa	Trunk type, where: aaa = CSA, TIE, or WAT.
TN	l s c u c u	Terminal Number. For Optiuon 11C.
NCOS	(0)-99	NCOS.
CLS	TLD	Toll Denied Class of Service.

LD 24 – Assign a DISA data block a Toll Denied and NCOS.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	DIS	DISA data block.
CUST	0-99 0-31	Customer number. For Option 11C.
SPWD	xxxx	Security password.
DN	xxx....x	DISA Directory number.
NCOS	(0)-99	NCOS.
COS	TLD	Toll Denied Class of Service.

LD 88 – Assign an Authorization code a Toll Denied and NCOS.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	AUB	Authorization code data block.
CUST	0-99 0-31	Customer number. For Option 11C.
SPWD	xxxx	Security password.
CLAS	(0)-115	Class code to be assigned.
NCOS	(0)-99	NCOS.
COS	TLD	Toll Denied Class of Service.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	All
Networking:	No

Night Key for Direct Inward Dialing (DID) Digit Manipulation

The Night Key for DID Digit Manipulation (NKDM) uses DID Incoming Digit Conversion (IDC) to convert received DID digits into a Night Service Directory Number (DN). NKDM is used to switch between Night and Day modes.

The Day/Night mode is controlled by a DID Route Control (DRC) key on an Attendant Console, or Meridian 1 proprietary telephone. There can only be one DRC key for each DID route.

The Night tree table is invoked in any of the following ways:

— when the attendant goes into Night Service, or the last attendant activates the POS BUSY key (provided that Attendant Overflow Position is not equipped)

— when an attendant activates the DID Route Control (DRC) key

— when a Console Presentation Group (CPD) attendant goes into Night Service, or

— when a Meridian 1 proprietary telephone activates the DRC key.

In each case, only the DID routes controlled by the initiating source (console or telephone) are affected.

Operating parameters

The maximum number of conversion tables per customer is 255. These tables are shared between the Incoming Digit Conversion (IDC) and the New Flexible Code Restriction (NFCR) trees.

The DID Route Control (DRC) key can only be configured on keys with lamp indicators.

When using the Night tree table, the same assumptions that apply to Incoming Digit Conversion (IDC) apply to this feature.

The Night tree table for DID Digit Manipulation (NKDM) applies only to DID routes.

For each DID route, there is only one configured DRC key per telephone.

For a Dialed Number Identification Service (DNIS) route, make sure that the correct table is selected for the conversion of incoming digits.

Feature interactions

Attendant Administration

The DID Route Control (DRC) key is not supported by Attendant Administration.

Attendant Overflow Position

When the last attendant activates the POS BUSY key, the system does not go into Night Service if an Attendant Overflow Position Directory Number (DN) is available.

Automatic Set Relocation

Delete the DRC key from a telephone before performing Automatic Set Relocation. If this is not done, the DRC lamp is activated on the wrong telephone.

Console Presentation Group Level Services

The Day/Night table can be activated with the DRC key by any attendant in the Console Presentation group.

Feature packaging

The Night Key for DID Digit Manipulation (NKDM) is part of base X11 system software. The following packages are required:

— Network Class of Service (NCOS) package 32

— New Flexible Code Restriction (NFCR) package 49, and

— Incoming Digit Conversion (IDC) package 113.

Feature implementation

LD 15 – Enable Incoming Digit Conversion for Night mode.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB FCR	Customer Data Block. Release 21 gate opener.
CUST	0-99 0-31	Customer number. For Option 11C.
NFCR	(NO) YES	Enable New Flexible Code Restriction.
- MAXT	1-255	Maximum number of NFCR trees.
IDCA	(NO) YES	Enable IDC. IDC cannot be disabled if any telephone has a DCR key.
- DCMX	1-255	Maximum number of IDC conversion tables. The sum of the values of MAXT and DCMX cannot exceed 255 per customer.

LD 49 – Add, change, or print code restriction trees.

Prompt	Response	Description
REQ	NEW CHG PRT	Create new, change, or print data.
TYPE	IDC	NFCR data block.
CUST	0-99 0-31	Customer number. For Option 11C.
DCNO	0-254	IDC tree number.
IDGT	0-9999 0-9999	Directory Number (DN) or range of DNs to be converted. The external DN to be converted is output and the user enters the internal DN. For example, to convert the external DN 3440 to 510, enter 3440. The system prompts 3440 and you enter 510.

LD 16 – Set IDC tree for Night mode. Note that a DID route cannot be removed if it is controlled by a DCR key.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
TKTP	DID	DID route.
IDC	(NO) YES	Enable IDC.
DCNO	0-254	IDC tree for Day mode.
NDNO	0-254 <CR>	IDC tree for Night mode. Set tree to the same number as Day mode (the default).

LD 12 – Define a DID Route Control (DRC) key on an Attendant Console.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT 1250 2250	Console type.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx DRC yyy	DRC, where: xx = key number 0-9 (0-19 on the M2250), and yyy = route number (0-511).

LD 11 – Define a DRC key on a Meridian 1 proprietary telephone.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx DRC yyy	DRC, where: xx = key number, and yyy = route number (0-511).

Feature operation

Follow these steps to change one DID route to Day/Night mode from the Attendant Console:

1 Select an idle loop key.

2 Press **DRC** and dial the access code of the DID route (ACOD).

If the DRC indicator is on steadily, the route is in Day mode.

If the DRC indicator is flashing, the route is in Night mode.

3 Press **DRC** again.

If the DRC indicator was on steadily, the route is put into Night mode.

If the DRC indicator was flashing, the route is put into Day mode.

Follow these steps to change all DID routes to Day/Night mode from the Attendant Console:

1 Select an idle loop key.

2 Press **DRC** and dial the octothorpe (#).

If the DRC indicator is on steadily, all routes are in Day mode.

If the DRC indicator is flashing, one or more routes are in Night mode.

3 Press **DRC** again.

If the DRC indicator was on steadily, all routes are put into Night mode.

If the DRC indicator was flashing, all routes are put into Day mode.

Note: To change from some routes in Night mode to all routes in Night mode, you must first put all routes into Day mode.

Follow these steps to change one DID route to Day/Night mode from a telephone:

1. Check the **DRC** indicator.

   If the DRC indicator is on steadily, the route is in Day mode.

   If the DRC indicator is flashing, the route is in Night mode.

2. Press **DRC**. The route toggles between Night and Day mode.

Night Restriction Classes of Service

The purpose of the Night Restriction Classes of Service (NRCLS) feature is to restrict the operation of the Call Waiting, Forced Camp-on, and Priority Override features so they operate during Night Service only. Therefore, the NRCLS feature will apply to any set which has Call Waiting, Forced Camp-on, or Priority Override features equipped.

Operating parameters

The Night Restriction Classes of Service (NRCLS) feature is available on any station.

Feature interactions

Call Waiting

If Call Waiting and Night Restriction for Call Waiting Class of Service (NRWA) are assigned, Call Waiting will be operational for the set only when Night Service is in effect.

Call Waiting Redirection

The Call Waiting Redirection feature applies to unanswered calls given Call Waiting treatment when the Night Restriction Classes of Service feature allows Call Waiting.

Camp-on, Forced

If Forced Camp-on and Night Restriction for Forced Camp-on Class of Service (NRCA) are assigned, Forced Camp-on will be operational for the set only when Night Service is in effect.

Override

If Priority Override and Night Restriction for Priority Override Class of Service (NROA) are assigned, Priority Override will be operational for the set only when Night Service is in effect.

Feature packaging

The Night Restriction Classes of Service feature is packaged under the Supplementary Features (SUPP) package 131.

Feature implementation

LD 10 and LD 11 – These overlays are modified to accept the following six new classes of service: NRCD, NRCA, NROD, NROA, and NRWD, NRWA.

Prompt	Response	Description
REQ	CHG NEW	Change, or add.
...		
CLS		Class of Service.
	(NRCD) NRCA	Night Restriction of forced Camp-on (Denied) Allowed. Forced Camp-on must be configured for the set. Assigning NRCD Class of Service allows Forced Camp-on to operate during both Night and Day Service. Assigning NRCA Class of Service restricts Forced Camp-on to operate during Night Service only. Default is NRCD.
	(NROD) NROA	Night Restriction of priority Override (Denied) Allowed. Priority Override must be configured for the set. Assigning NROD Class of Service allows Priority Override to operate during both Night and Day Service. Assigning NROA Class of Service restricts Priority Override to operate during Night Service only. Default is NROD.

	(NRWD), NRWA	Night Restriction of call Waiting (Denied) Allowed. Call Waiting must be configured for the set. Assigning NRWD Class of Service allows Call Waiting to operate during both Night and Day Service. Assigning NRWA Class of Service restricts Call Waiting to operate during Night Service only. Default is NRWD.

LD 20 – This overlay is modified to print the six new Classes of Service when the Terminal Number block is printed.

LD 81 – This overlay prints DES to TN and last service change information for selected features. The classes of service NRCA, NRCD, NROA, NROD, NRWA, and NRWD are now allowed.

Prompt	Response	Description
REQ	LST CNT END	List telephones equipped with the feature specified by the prompt FEAT. Print a count of telephones equipped with the feature specified by the prompt FEAT. End overlay activity.
CUST	0-99 0-31	Customer number. For Option 11C.
DATE	1-31 Jan-Dec ACT <CR>	Print data from activity date specified. Print data from last activity date. Disregard date restrictions.
PAGE	(NO), YES	Print data on a per page basis.
DES	XXXXXX X+ + <CR>	Print station with designator XXXXXX. Print data for stations with designators starting X. Print data for all stations with no designator. Print data for all stations with designators.
FEAT	NRCA, NRCD NROA, NROD NRWA, NRWD	Night Restriction of Forced Camp-on Allowed, or Denied. Night Restriction of Priority Override Allowed, 0r Denied. Night Restriction of Call Waiting Allowed, or Denied.

LD 83 – This overlay is modified to print the NRWA, NRWD, NRCA, NRCD, NROA, and NROD classes of service for sets when the Supplementary Features (SUPP) package 131 is equipped.

Feature operation

A customer or a Console Presentation Group (CPG) can be put into Night Service manually by pressing the Night key on the Attendant Console or automatically by Scheduled Access Restriction (SAR) or Attendant Forward No Answer (AFNA).

Depending on the Class of Service (CLS) and key assignments, the operation of the features will be allowed or denied as summarized in the following table:

CLS	Feature X Allowed	Feature X Denied
NRXA	Feature **X** is restricted to operate during Night Service only.	Feature **X** always denied.
NRXD	Feature **X** operates whether Night Service is active or not.	Feature **X** always denied.

Legend:
NR**X**A: Night Restriction of feature **X** Allowed for this set.
NR**X**D: Night Restriction of feature **X** Denied for this set.

Where **X** =:

W for Call Waiting
C for Forced Camp-on, or
O for Priority Override.

Night Service

Night Service permits incoming calls normally directed to the attendant to be routed to a defined destination. A separate Night key/lamp pair allows the attendant to put the system into Night Service.

Three types of Night Service are provided which the customer can specify separately or in any combination:

— Selected Trunks to Selected Directory Number (DNs): Some or all of the trunks can be assigned to ring selected DNs when the system is in Night Service. The assignment of trunks to stations can be modified by the attendant or by a service change.

— Night Answer Telephone: All calls normally routed to the Attendant Console can be routed to one particular DN that is designated as the night answer destination for the customer. Trunk Answer From Any Station (TAFAS) can be used to pick up calls routed to this number. With TAFAS in effect, incoming calls activate a common alerting device, such as a bell, when the system is in Night Service. Any user can answer the call by dialing the Special Prefix (SPRE) code and then pressing 4.

— Night Service by Time of Day (NSTD): Available in X11 Release 12 and later, NSTD allows one of a group of Directory Numbers (DNs) to be selected for call routing based on the time of day instead of all calls being routed to a fixed Night Service DN. NSTD allows the definition of up to four Night DNs with a time associated with each. Calls are forwarded to the appropriate DN by the associated time.

Operating parameters

Night Service can only be activated from the Attendant Console.

Any restrictions or features assigned to the night answering station apply. Therefore, a fully restricted (FRE) Class of Service should not be used for Night Service Directory Numbers (DNs), unless the FRPT prompt in LD 17 is OLFR (allow FRE telephones to serve as a Night DN).

A bell circuit or alerting device must be provided by the customer for TAFAS. This device must be compatible with the 20 Hz ringing signal (i.e., two seconds on, four seconds off).

If a trunk is assigned a Night DN other than the Night Answer Number defined in the Customer Data Block, incoming calls to that trunk cannot be picked up with the TAFAS feature. Assignment in LD 14 takes precedence over the Customer Data Block.

If an attendant is not assigned to a customer, the customer is automatically in Night Service upon system start-up. The following tables show how calls are directed during Night Service, depending on the time of day:

Call is directed to Night DN	Between times:
NIT1 DN	TIM1 and TIM2
NIT2 DN	TIM2 and TIM3
NIT3 DN	TIM3 and TIM4
NIT4 DN	TIM4 and TIM1

It is possible to remove a defined night DN without modifying the other DNs. For example, if NIT3 is removed, calls are directed as follows:

Call is directed to Night DN	Between times:
NIT1 DN	TIM1 and TIM2
NIT2 DN	TIM2 and TIM4
NIT4 DN	TIM4 and TIM1

Feature interactions

Attendant Overflow Position

A call rerouted through the Attendant Overflow Position feature is not redirected to the Night DN if the system is subsequently put into Night Service. When all Attendant Consoles are in Position Busy, the system will not go into Night Service until the AOP Busy key is activated.

Deactivating the AOP Busy key after the system has been placed in Night Service does not affect the Night Service feature.

Attendant Position Busy

When the last console operator activates the Position Busy key or the Night key, Night Service is put into effect. Incoming calls receive the customer-specified night treatment.

Automatic Wake Up

Unanswered Automatic Wake Up calls going through Attendant Recall are discarded if the Attendant Console is in the Night Service mode. Automatic Wake Up may still be programmed when the Attendant Console is in Night Service.

Call Forward Busy

When the system is in Night Service, Direct Inward Dialing calls forwarded by Call Forward Busy are routed to the specified night number. If the night telephone is busy, subsequent calls receive busy tone.

Call Pickup Network Wide

The Call Pickup Network Wide feature can be used to pick up a call to the night number if it is ringing an ordinary station (that is, analog (500/2500 type) telephone, 16-button Dual-tone Multifrequency, or proprietary set).

Call Waiting Redirection

Night Service has the same interaction with the Call Waiting Redirection feature as attendant-extended calls. Since the Call Waiting Redirection feature applies CFNA treatment to a Call Waiting call, the Call Waiting Redirection feature also has precedence over the Call Waiting recall timer.

Calls Waiting in Attendant Queue

Incoming calls ringing at the Attendant Console at time changeover are routed to the Night DN that just expired. New calls are routed to the new Night DN. If the attendant cancels Night Service, new calls are presented to the Attendant Console.

Once a call begins ringing at a Night DN, it stays there even if Night Service is cancelled or the timer expires.

Departmental Listed Directory Number

Departmental Listed Directory Number does not affect Night Service (including TAFAS). Calls presented to the LDN from an external source will queue for the night bell. All other attendant calls receive busy treatment if the night Directory Number (DN) is busy.

Directory Number Expansion

If the Directory Number Expansion (DNPX) package is equipped, the Night DNs can be up to seven digits; otherwise, the DN can be a maximum of four digits.

Distinctive/New Distinctive Ringing

Incoming calls terminating on a night Directory Number (DN) ring distinctively.

DPSNN1 Diversion

If a diverted call encounters an attendant in night service, the call receives Night Service Diversion if available.

End-to-End Signaling

Night Service works together with Attendant End-to-End Signaling (AEES). However, do not press this feature key while using AEES, or the Dual-tone Multifrequency (DTMF) code signals may be blocked.

Equi-distribution Network Attendant Service Routing

When the attendant goes into Night Service, calls presented to the attendant receive NAS routing in an attempt to reach another attendant that is in day service, rather than being routed to the local night DN.

Manual Line Service

When the system is in Night Service (NSVC) mode, all telephones with a manual Class of Service are routed to the telephone designated as the night number for the customer group.

Meridian 911 Call Abandon

Abandoned calls can be forwarded to the Night Call Forward DN if the Night Forward DN is an ACD DN. If a primary answering center goes into Night Service while there are abandoned calls in the queue, those abandoned calls are dropped. A CDR N record is printed if CDR is configured.

Multi-Party Operations

During Night Service, mishandled calls are routed to the night DN. External calls, other than DID calls, are queued until answered. TIE calls are disconnected if the night DN is busy.

Night Service

If the system is in Night Service mode, mishandled calls which are routed to the attendant are rerouted to the appropriate Night Service DN. External trunk calls, other than DID, are queued till they are answered.

TIE trunk calls are not queued at the Night Service DN. If the Night Service DN is busy, TIE calls are disconnected.

Night Service Enhancements

When the Night Service key is pressed on any Attendant Console, the customer enters Night Service and all Attendant Consoles are made Position Busy. It is then necessary to check all consoles for presented but unanswered calls, which must be cleared and requeued.

Recorded Overflow Announcement

The Recorded Overflow Announcement feature is inactive when the system is in Night Service.

Series Call

If the attendant extends a Series Call and goes into Night Service before it recalls to the attendant, the call recalls to the night DN and Series Call treatment is cancelled.

Trunk to Trunk Connection

If an attendant is placed in Night Service, calls to the attendant are directed to a station with the Night DN. Recalls are not directed to the Night DN. Recalls are put in the attendant call waiting queue when in Night Service.

The following interactions apply to Night Service by Time of Day (NSTD):

Call Park Recall

Calls parked by the attendant recall on the Night Service DN that is current at the time of recall.

Multi-Tenant Night Service

The same conditions that apply to the customer night number also apply to the Multi-Tenant Night Service. In X11 Release 15 and later software, Console Presentation Group (CPG) allows separate night treatment for each tenant.

Trunk Answer from Any Station

When a DN changeover occurs while an incoming call is ringing the current Night DN and a new incoming call is ringing the new Night DN, a user activating Trunk Answer from Any Station (TAFAS) picks up the call from the Night DN that just expired. However, if the ringing call is not picked up within one minute after the Night DN time changeover, the user can no longer pick up the call using TAFAS.

Feature packaging

Night Service is included in base X11 system software.

Feature implementation

LD 15 – Add or change Night Service for a customer.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB NIT	Customer Data Block. Release 21 gate opener.
CUST	0-99 0-31	Customer number. For Option 11C.
NITE	xxx...x, X	Night Service DN (prior to X11 Release 12 only).
- NIT1	xxx...x, X	Night Service DN 1 (enter X to remove). Night Service DN times must be defined in ascending order.
- TIM1	0-23 0-59	DN 1 time (hour and minute).
- NIT2	xxx...x, X	Night Service DN 2 (enter X to remove).
- TIM2	0-23 0-59	DN 2 time (hour and minute).
- NIT3	xxx...x, X	Night Service DN 3 (enter X to remove).
- TIM3	0-23 0-59	DN 3 time (hour and minute).
- NIT4	xxx...x, X	Night Service DN 4 (enter X to remove).
- TIM4	0-23 0-59	DN 4 time (hour and minute).

LD 14 – Add or change Night Service DN for trunks.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	COT	Trunk type.
TN	l s c u c u	Terminal Number. For Option 11C.
NITE	xxx...x, X	Night Service DN for this trunk (enter X to remove).

Feature operation

To place a customer into Night Service:

— Press **Shift** plus  at any console, or unplug all handsets and headsets.

To cancel Night Service when all handsets and headsets are unplugged:

— Plug in at least one handset or headset.

To cancel Night Service at a console when a handset or headset is plugged in:

— Press **Shift** plus 

Note: If all Attendant Consoles are put in Position Busy, the system automatically goes into Night Service.

Introduced in X11 Release:	14.46E
Networking:	No

Night Service Enhancements

This enhancement introduces the following capabilities:

— All Calls Remain Queued for Night Service,

— Recall to Night DN,

— Requeuing of Attendant Presented Calls, and

— Camp-on from Inquiry Call (Station Camp-on).

All Calls Remain Queued for Night Service

This capability ensures that when Night Service is activated all calls in the attendant queue remain queued for Night Service treatment. Depending on the call type, the call may be presented to the Night DN, or continue waiting for the called party to answer. This includes Call Forward No Answer calls, recalls, and transfers to the attendant.

This capability applies to both standalone and networking environments. Within a networking environment, if Network Attendant Service (NAS) is equipped at all nodes, the calls are presented to a remote attendant, remote Night DN, or local Night DN, depending on the NAS configuration. This treatment applies to external calls only, since internal calls are not queued against a remote Night DN. If NAS routing is not involved, external calls are presented to the local Night DN.

Recall to Night DN

If the attendant camps-on party A to a busy set B, then goes into Night Service, the recall goes to the Night DN only if A is an external party (that is, CO, DID, FEX, WATS). This happens for a local camp-on.

For a Meridian Customer Defined Network (MCDN) camp-on with A at the far end of the MCDN NAS network and for a DPNSS1 camp-on with A at the far end of the DPNSS1 network the situation is as follows. If A is an internal party, the recall is left in the attendant queue, and can be answered by the attendant if the attendant returns to day service.

This internal/external difference does not hold true if the International Supplementary Features (SUPP) package 131 is equipped.

Requeuing of Attendant Presented Calls

The Requeuing of Attendant Presented Calls is an enhancement to the Attendant Forward No Answer feature. If a call presented to an Attendant Console is not answered, pressing the Position Busy key causes the call to be placed in the attendant queue.

If the console is the customer's last-active console, and Attendant Overflow Position (AOP) is active, a ringing call or a Call Waiting recall on the Destination side is disconnected. This ensures that any queued call will be presented at the AOP.

Any call presented at the AOP is not removed from the console and requeued if the Position Busy key is pressed.

The call is removed unanswered only if the Attendant Forward No Answer feature is active. In this case, after the Attendant Forward No Answer time out expires, the call is requeued and the AOP is idled.

All consoles will enter the Position Busy state if the Night Service key is pressed on any of the customer's consoles. Therefore, all consoles should be checked for presented, but unanswered calls, which have been requeued.

Camp-on from Inquiry Call (Station Camp-on)

With this feature, any internal station can camp an external call on to another internal station that is busy. Prior to the introduction of this feature attendant's were the only parties that could camp calls on to busy internal stations. The term internal station includes stations on other nodes within an Meridian Customer Defined Integrated Services Digital Network (MCDN).

When a transferring party reaches a busy desired internal party, the transferring telephone will receive ringback tone (providing certain conditions are met). When the transferring party completes the transfer, the external (calling) party will Camp-on to the desired party and the external party (an external party is any CO, DID, FEX, or WATS call) will receive ringback tone or announcement.

This feature applies to both standalone and network environments.

Within a network environment, the transferring and Camped-on to stations may be on the same or different nodes, as long as all nodes are configured with Network Station Camp-on.

Operating parameters

Camp-on from Inquiry Call (Station Camp-on)

The restrictions which currently apply to the operation of the Camp-on feature from an Attendant Console also apply to Camp-on from Inquiry Call (Station Camp-on).

These restrictions are:

— Camp-on will not be permitted if the desired station is in a state other than established (for instance, ringing or dialing).

— Only one call at a time may be Camp-on a busy station.

— Calls cannot Camp-on to a station with the Call Waiting feature configured.

— The station camped-on to will be given Warning Tone only if the customer has Camp-on Tone Allowed (CTA) in the Customer Data Block (LD 15) and the station has Warning Tone Allowed (WTA) Class of Service assigned. If the station has Warning Tone Denied (WTD) Class of Service assigned the Camp-on will take effect without giving any Camp-on Tone to the camped-on to (desired) party.

— The transferring station will receive Busy Tone only if the response to the STCB prompt in the Customer Data Block (LD 15) of the Camped-on to (desired) set is YES. Otherwise, the transferring station will receive ringback tone.

Camp-on Indication

When a call is extended from an attendant to a busy station there is a specific combination of tones and indicator states to identify the Camp-on state.

When an inquiry call is made from a station, there is only one way for the user to distinguish between a busy set and an idle ringing set. That way is to ensure that the response to the STCB prompt in the Customer Data Block (LD 15) of the Camped-on to (desired) set is YES. Otherwise, ringback tone is provided in both cases.

Night DN

When the customer goes into Night Service, if the Night DN is idle, only the first call is presented to it.

The Night DN may be defined as a multiple appearance DN with multiple call arrangement; all sets assigned the Night DN should be on the same node.

According to NAS routing, the Night DN defined on a node must be on the given node (local). If for any reason the Night DN is not on the local node Night Service Enhancements (NSE) are no longer supported.

In any case, NAS routing takes precedence over NSE, so if NAS routing is involved the call will be presented to the Night DN defined according to the NAS configuration.

If NAS routing is not involved and the Night DN defined on this node is located at a remote node (NSE no longer supported), the Night DN must be a remote Attendant DN to ensure calls are queued.

Night Service Network Environment

In network configurations with NAS routing, the Night Service Enhancements feature must be configured on each node in the network.

Feature interactions

Attendant Clearing during Night Service

The Night Service Enhancement features take precedence over Attendant Clearing during Night Service.

Attendant Forward No Answer

Any call which has been presented to the Attendant Overflow Position cannot be not be removed from the set and requeued by pressing by Make Set Busy (MSB) key. The call will only be removed if the Attendant Forward No Answer feature is active, and the Attendant Forward No Answer Timer has timed out. In this case, the call is requeued and the Attendant Overflow Position is idled.

Attendant Interpositional Transfer

The requeuing of interpositional calls is not allowed. Night Service enhancements do not apply to interpositional calls, which remain on the console until answered.

Attendant Overflow Position

If a call with a ringing party on the destination side is presented at the last-active Attendant Console, and there is an active Attendant Overflow Position, then the ringing destination will be disconnected when the call is requeued. Likewise, if the call is a Call Waiting recall, Call Waiting will be canceled.

Attendant Position Busy

Any call that has been presented to the Attendant Overflow Position cannot be removed from the console and requeued by pressing the Make Set Busy (MSB) key. The call will be removed only if the Attendant Forward No Answer feature is active and the Attendant Forward No Answer Timer has timed out. In this case, the call is requeued and the Attendant Overflow Position is idled.

Call Forward All Calls

Any call which has been presented to the Attendant Overflow Position cannot be removed from the console and requeued by pressing the Make Set Busy (MSB) key. The call will only be removed if the Attendant Forward No Answer feature is active, and the Attendant Forward No Answer Timer has timed out. In this case, the call is requeued and the Attendant Overflow Position is idled.

Call Waiting

Call Waiting will be applied by Night Service Enhancements to terminate incoming Night calls to busy Night DNs. This will still be done even if the Night DN is an analog (500/2500 type) telephone with Call Waiting Denied (CWD) Class of Service, or if the Night DN is a Meridian 1 proprietary telephone without a Call Waiting (CWT) key assigned.

All telephones will be given Night Call Waiting tone, if the NWT prompt in overlay 15 was responded to with "YES," regardless of the Warning Tone (WTA/WTD) Class of Service setting of the telephone. Meridian 1 proprietary telephones will be given Night Call Waiting tone in the handset instead of the speaker buzz given for Call Waiting.

Call Waiting Redirection

Night Service has the same interaction with the Call Waiting Redirection feature as attendant-extended calls. Since the Call Waiting Redirection feature applies CFNA treatment to a Call Waiting call, the Call Waiting Redirection feature also has precedence over the Call Waiting recall timer.

Centralized Attendant Service

Centralized Attendant Service (CAS) takes precedence over Night Service. If a user in a remote node in Night Service deactivates CAS and Camps-on an external call from the night station to a busy DN, and then reactivates CAS, any subsequent Camp-on recalls will be routed to the remote DN.

Dial Impulse Analog (500/2500 type) Telephone

A Dial Impulse analog (500/2500 type) telephone station must have TSA Class of Service to perform a station Camp-on.

Direct Inward System Access

It is not possible to assign a Night Service Group Number to any trunk that is a member of a route that is set to auto-terminate on a Direct Inward System Access DN.

Interposition Attendant Calls

This enhancement does not apply to interposition calls, which remain on the console until answered. The requeuing of interpositional calls is not allowed.

Network Attendant Service

Network Attendant Service (NAS) is mutually exclusive with Centralized Attendant Service and Attendant Overflow Position. The routing configuration for NAS will apply during Night Service. External calls and recalls may be queued to a remote Night DN, if defined. Internal calls and internal recalls queued during Day Service will be dropped, if the Night DN has been defined on a remote node.

For Camp-on from Inquiry Calls, NAS must be equipped at each node of the network.

Night Service

When the Night Service key is pressed on any Attendant Console, the customer enters Night Service and all Attendant Consoles are made Position Busy. It is then necessary to check all consoles for presented, but unanswered calls which must be cleared and requeued.

Recall with Priority during Night Service, Network Wide

If Recall with Priority during Night Service is equipped along with either the Night Service Improvement or Enhanced Night Service feature, calls are processed according to priority.

Trunk to Trunk Connection

Recalls made while the attendant is in Night Service are routed to the Night DN, if the original call is an external call. In such a case, the destination party is disconnected, the internal network trunk is released and the original extended call is presented to the Night DN. If the original call is internal, recalls are put in the attendant call waiting queue when in Night Service.

Feature packaging

The All Calls Remain Queued for Night Service, Recall to Night DN, and Requeuing of Attendant Presented Calls Night Service Enhancements are packaged as part of the International Supplementary Features (SUPP) package 131 for standalone applications. For network applications, the requirements are the International Supplementary Features (SUPP) package 131 and the Network Attendant Service (NAS) package 159 and its prerequisites.

For standalone Camp-on from Inquiry Call (Station Camp-on) applications the requirements are the Station Camp-on (SCMP) package 121 and the International Supplementary Features (SUPP) package 131.

For network Camp-on from Inquiry Call (Station Camp-on) applications the requirements are the Station Camp-on (SCMP) package 121, the International Supplementary Features (SUPP) package 131 and the Network Attendant Service (NAS) package 159 and its prerequisites.

Feature implementation

LD 15 – This overlay is modified to prompt and accept responses to STCB (Station Camp-on Busy Tone) and NSCP (Network Station Camp-on). In response to the STCB prompt, enter YES or NO to allow or deny Station Camp-on Busy Tone. In response to the NSCP prompt, enter YES or NO to allow or deny Network Station Camp-on on a particular node.

The prompt STCB will be output only if the SCMP (121) package is equipped. The prompt NSCP will be output only if the SCMP (121) and NAS (159) packages are equipped. By default, these two prompts will be set to NO.

Prompt	Response	Description
REQ	CHG NEW	Change, or add.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
...		

- STCB	(NO), YES	Station Camp-on Busy tone. Enter NO if Busy Tone is not to given to the transferring (controlling) party when the desired station is busy. Enter YES if Busy Tone is to be given to the transferring (controlling) party when the desired station is busy. The default is NO.
- NSCP	(NO), YES	Network Station Camp-on. Enter NO if sets on this node are not allowed to have calls camped-on by sets in other nodes. Enter YES if sets on this node are allowed to have calls camped-on by sets in other nodes. The default is NO.

LD 21 – This overlay is modified to print the STCB and NSCP prompts and their responses when the Customer Data Block is printed.

The STCB prompt and its response will be output only if the SCMP (121) package is equipped. The NSCP prompt and its response will be output only if the SCMP (121) and NAS (159) packages are equipped.

LD 22 – This overlay is modified to print the SCMP package mnemonic if the Station Camp-on package (121) is equipped.

Feature operation

Prior to describing the operation of the Night Service Enhancements the following terms are defined in terms of these feature operations:

Night DN

The Night DN pertains to Night DNs defined on a customer basis.

According to NAS routing the Night DN defined on a node must be on the given node (local).

External Call

Any call originated by the Public Switched Telephone Network (PSTN) is said to be an external call. This includes the following cases:

— Calls originating on a Public Exchange (Central Office [CO]), Foreign Public Exchange (FEX), Direct Inward Dial (DID), or Wide-area Telephone Service (WATS) trunk and terminating on the local node, and

— Calls originating on a CO, FEX, DID, or WATS trunk on a remote node, Integrated Services Digital Network (ISDN) TIE trunks, and NAS routed Public Switched Telephone Network (PSTN) or ISDN TIE trunks which are handled at the NAS node.

Non ISDN TIE trunks (local and remote) are said to be private trunks and are not treated as carrying external calls, although we may have a PSTN call involved at the originating node.

This definition includes both the standalone and network cases.

Requeuing of Attendant Presented Calls

Prior to the introduction of the Requeuing of Attendant Presented Calls, when a call had been presented to an Attendant Console it remained presented on the console, even if the Position Busy key was pressed.

The Requeuing of Attendant Presented Calls capability changes the system operation such that, if the Position Busy key is pressed on the console when an unanswered call has been presented to it the call will be returned to the attendant queue as if an AFNA time out had occurred.

This capability will not apply if the call is an interposition attendant (attendant to attendant) call. In this case, the call will remain on the console until answered.

In cases where the console is the last active console of the customer and there is an active AOP, if the call involves a ringing party on the destination side, the ringing will be disconnected. Similarly if the call is a Call Waiting recall, the Call Waiting will be canceled. This ensures that the required call will be presented on the AOP, irrespective of normal call type restrictions.

Note that all consoles will enter the Position Busy state if the Night Service key is pressed on any one of a customer's Attendant Consoles. In this case, all consoles must be checked for presented, but unanswered calls which must be cleared from the console and requeued.

Call Handling in Night Service

Calls already Queued when Night Service is Entered

Standalone case

Any external call which is queued, waiting to be serviced by an Attendant Console, when a customer goes into Night Service will continue to be queued until it can be presented to the appropriate Night DN.

Network case

As NAS takes precedence over NSE, if NAS routing is involved, the call will be presented to a remote attendant, or remote Night DN, or local Night DN, according to the NAS configuration.

If NAS routing is not involved, the call will be presented to local Night DN.

External Calls already Queued when Night Service is Entered

Operation Prior to Night Service Enhancements

The treatment of queued external calls was as follows:

— Dial "0" calls from DIDs or incoming CO calls remained queued for the Night DN.

— Call Forward Busy calls remained queued for the Night DN.

— Call Forward No Answer calls were not queued for a busy Night DN. If a call could not be presented immediately it was removed from the queue and the originating party was given Busy Tone.

— Attendant Recalls (ARC) and transfers to the attendant DN were removed from the attendant queue. The consultation call was "canceled", if the held call was an external party it was reconnected to the transferring (controlling) party.

— All intercepts involving an external party were queued for the Night DN.

— Timed reminder recalls remained queued, but were not presented to the Night DN.

Operation with Night Service Enhancements

The NSE capabilities change the operation such that Call Forward No Answer calls, ARCs, and transfers to the attendant will remain queued for the Night DN. In addition to these call types, timed reminder recalls will also be presented to the appropriate night DN.

Timed reminder recalls treatment is the following:

— Ringing stops for slow answer recalls when the recall occurs.

— Call Waiting is canceled when the recall occurs.

— Camp-on is canceled when the recall occurs.

Internal Calls already Queued when Night Service is Entered

Operation Prior to Night Service Enhancements

Any internal call that was already queued for the attendant was not queued for the Night DN.

When a customer went into Night Service, if the Night DN was idle, the first call was presented to the Night DN. Any internal calls not presented in this way were given busy tone and removed from the queue.

Operation with Night Service Enhancements

Standalone case

With NSE the operation is changed such that all internal calls which should be presented to the Night DN will remain queued until the customer Night DN becomes available.

Network case

If the call was extended by the attendant over DPNSS1 or MCDN with NAS active, and the call is camped-on or call waiting at the remote node, the call will remain queued at the local node waiting for an answer at the remote node.

Timed Reminder Time Outs during Night Service

When a timed reminder time out occurs during Night Service, depending on the call type, the call may be presented to the Night DN or continue waiting for the called party to answer. External (PSTN originated) calls will be presented to the Night DN or, if the Night DN is busy will wait in the queue until the Night DN becomes available.

In the case of a timed reminder Camp-on recall, the Camp-on is canceled when the recall occurs (time out).

In case of a slow answer recall, the desired set will be disconnected when the recall occurs (time out).In case of a timed reminder Call Waiting recall, the Call Waiting will be canceled when the recall occurs (time out).

According to NAS routing these calls may be presented to a remote attendant or a remote Night DN. When the NAS routing starts, the destination (desired party) is released and the call is presented or queued to the appropriate terminal (i.e., remote attendant or local Night DN or remote Night DN).

External calls that recall will be presented to, or queued for, the Night DN.

Internal calls that recall will be dropped when NAS routing is involved and the Night DN is at a remote node, because when NAS routing takes place internal call recalls are not queued for the Night DN.The station to which the call is being transferred (i.e., the station on which the call is ringing, Call Waiting or camped-on) does not have to be located on the same node as the transferring (controlling) station.

If the attendant on the same node as the Night DN comes back to Day Service, timed recalls queued for the Night DN will be presented to the attendant as recalls.

Camp-on from Inquiry Call (Station Camp-on)

Standalone case

Any station, not necessarily the Night DN, attempting to transfer an external call, may, during the associated inquiry call, camp the trunk on to a busy station.

The camp-on will take affect from the moment the transferring station has completed the transfer to the desired DN.

The transferring station will hear Ringback Tone or Busy Tone depending on the option entered in response to the STCB prompt in the Customer Data Block (LD 15). This prompt applies to any set, not just the Night DN. By default (STCB is set to NO), the transferring party will hear Ringback Tone. The desired station will hear Camp-on tone if it has WTA Class of Service assigned. Otherwise, if it has WTD Class of Service, the Camp-on will take affect without the desired party being informed a call is camped-on.

When the transfer is completed, the external party is camped-on to the desired station and receives either ringback tone or an announcement.

Network case

Any station, not necessarily the Night DN, attempting to transfer an external call across an ISDN network may, during the associated inquiry call, Camp-on the trunk on to a busy station.

The location of the transferring party has no effect on the Station Camp-on capability.

The Camp-on will take Affect from the moment the transferring station has completed the transfer to the desired DN.

The transferring station will hear ringback tone or busy tone depending on the option entered in response to the STCB prompt in the Customer Data Block (LD 15). This prompt applies to any set, not just the Night DN. By default (STCB is set to NO), the transferring party will hear ringback tone. The tone given, either ringback tone or busy tone, is determined by the node in which the desired (Camped-on to) party resides.

The desired station will hear Camp-on tone if it has WTA Class of Service assigned. If it has WTD Class of Service, the Camp-on will take affect without the desired party being informed a call is camped-on.

When the transfer is completed, the external party is camped-on to the desired station and receives either ringback tone or an announcement.

Recall Timing on Camp-on Calls

When any station extends an external call, recall timing will be initiated if the call is camped-on to a busy station.

The recall timing will start from the moment that the extending station "releases" the call. The value of the recall timer is set by the prompt RTIM in the Customer Data Block (LD 15).

At the recall, the Camped-on call will be routed to the attendant. If the attendant is in Night Service, night treatment is given, and if NAS routing is active, the call will be routed according to the NAS configuration.

Standalone case

When the recall to the attendant occurs, the Camp-on is canceled. If the attendant is busy during the recall, the recall will be queued.

Network case

When the recall occurs and the attendant has answered the recall, the call will still be camped-on to the desired party. If during the recall the attendant is busy, the recall will be queued.

Night Service, Enhanced (X11 Release 20)

This feature modifies the existing Night Service operation by allowing Public Network (Central Office [CO], Direct Inward Dialing [DID], Foreign Exchange [FEX], and Wide Area Telephone Service [WATS]) trunks to be assigned to specific Directory Numbers (DN) during Night Service.

With this feature each customer will be able to assign Public Network trunks to one of nine Night Groups. Each Night Group will allow the customer to define up to nine Night DNs. During Night Service, incoming calls will be routed to one of the Night DNs defined for the group. The actual DN the call will be routed to is determined by the Night Service Option number selected at that time.

The customer will also be able to define whether Night Call Waiting tone will be given to Night stations. With Night Call Waiting tone allowed, busy Night stations are notified when an incoming call is terminating on them. The incoming call will be queued on the Night station until it becomes idle. When the Night station becomes idle, the incoming call will be presented.

This enhancement allows incoming DID trunks to be queued against busy Night stations, thereby making their operation the same as all other Public Network trunks.

Normal Night Service

With the feature active, the existing Night Service feature is enhanced by providing a night (NITE) prompt for DID trunks. Night numbers for DID trunks can be defined in their respective trunk blocks against the prompt. Attendants will be able to change their night numbers by specifying their corresponding access codes and member numbers using the existing Night Service feature.

Group Night Service

The customer is allowed to assign individual Public Network trunks to one of nine Night Group numbers (1 to 9). Each Night Group has up to nine Night Directory Numbers associated with it. During Night Service, incoming calls on a trunk will be routed to one of the Directory Numbers associated with that trunk. The actual number called is determined by a Night Service Option number corresponding to the Night Group number programmed by the attendant during Day service.

When an incoming call is routed to a busy directory number, an optional Night Call Waiting tone may be applied to that number to notify the user that a call is waiting. The call on the trunk will be queued until the night directory number becomes free.

Operating parameters

The same feature requirements apply as for Night Service.

Enhanced Night Service does not apply to auto-terminate trunks.

Enhanced Night Service is permanently activated if the system has no attendant and the ENS option is set to "YES." In this case, the Night Service Option number can only be programmed in the Customer Data Block (LD 15).

Enhanced Night Service uses one Speed Call list as the Night Number Table.

The operation of the optional Night Call Waiting Tone is the same as Call Waiting Tone.

Night Service Option 0 and Night Service Group 0 are reserved for the customer Night number and should not be programmed in LD 18.

Feature interactions

AC15 Recall: Timed Reminder Recall

The Night Service Enhancements feature is used to direct the call to the Night DN if the original call is an external call and the SUPP package 131 is equipped. When there is an AC15 recall and the attendant is in Night Service, the called party is disconnected (the AC15 trunk is released) and the original call is presented to the Night DN.

Call Waiting (CWT)

This feature will terminate incoming Night calls to busy Night DNs by applying Call Waiting. This will still be done even if the Night DN is an analog (500/2500 type) telephone with Call Waiting Denied (CWD) Class of Service, or if the Night DN is a Meridian 1 proprietary telephone without a Call Waiting (CWT) key assigned.

All telephones – both analog (500/2500 type) telephones and Meridian 1 proprietary telephones – will be given Night Call Waiting tone, if the NWT prompt in LD 15 was responded to with "YES," regardless of the Warning Tone (WTA/WTD) Class of Service setting of the telephone. Meridian 1 proprietary telephones will be given Night Call Waiting tone in the handset, instead of the speaker buzz given for Call Waiting.

Direct Inward System Access (DISA)

It is not possible to assign a Night Service Group Number to any trunk that is a member of a route that is set to auto-terminate on a DISA DN.

Multi-Party Operations

During Night Service, mishandled calls are routed to the night DN. External calls, other than DID calls, are queued until answered. TIE calls are disconnected if the night DN is busy.

Multi-Tenant service

Any restrictions that exist in the system preventing individual Tenant access to certain routes will not be checked when the Night Number Table is programmed. It will be up to the technician to ensure all such restrictions are taken into consideration.

The tenant to route restrictions will be enforced when an attempt is made to terminate an incoming call on a Night DN via the Night Number Table. If the termination to the Night DN is not allowed, Overflow tone (Fast Busy) will be given to the incoming trunk.

Trunk Barring (Telephones)

Any incoming trunk call that is routed by Enhanced Night Service to a telephone from which it is barred will not be connected. Overflow tone (Fast Busy) will be given to the incoming trunk instead.

Trunk to Trunk Barring

Any incoming trunk call that is routed to an outgoing Public Network trunk will be barred if Enhanced Night Service is active. Overflow tone (Fast Busy) will be given to the incoming trunk instead. This restriction is in addition to the configured Trunk Barring for the system.

Warning Tone

All telephones – both analog (500/2500 type) telephones and Meridian 1 proprietary telephones – will be given Night Call Waiting tone, if the NWT prompt in LD 15 was responded to with "YES," regardless of the Warning Tone (WTA/WTD) Class of Service setting of the telephone.

Feature packaging

Enhanced Night Service (ENS) is packaged as package 133.

Feature implementation

LD 18 – Configure Night Number Table.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	SCL	Speed Call List number
LSNO	xxx	List number. Enter list number; this number will be entered in response to the NNT prompt in LD 15 (Customer Data Block).
DNSZ	xx	Enter maximum excepted length required.
SIZE	100	Enter 100 to ensure that definitions for Options 1-9 and Groups 1-9 may be input.
STOR	xy z...z	Define Night Number Table entry, where: x is the Night Service Option number (1-9) y is the Night Service Group number (1-9), and z...z is the DN to which calls will be routed. This must be a valid station DN within the system. Network Access Codes are not allowed. ***Note:*** Night Service Option 0 and Night Service Group 0 are reserved for the customer Night number and should not be programed, (i.e., 00, 01, 02, 03, 04, 05, 06, 07, 08, 09, 10, 20, 30, 40, 50, 60, 70, 80, and 90).

LD 15 – Configure Enhanced Night Service.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	CDB	Customer data block.
...		
ENS	(NO) YES	(Disable) enable Enhanced Night Service.
- NWT	(NO) YES	(Disable) enable Night Call Waiting tone.
- NNT	0-253	Enter the Speed Call List (LSNO) number of the Night Number Table defined in LD 18.
- NSO	0-9	Night Service Option number.

LD 14 – Configure Enhanced Night Service for trunks.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DID	Direct Inward Dial.
...		
NGRP	(0)-9	Night Service Group number.

Feature operation

Night number assignment from Night Number Table

A Speed Call List (SCL) is specified in the Customer Data Block (CDB), LD 15, for the purpose of storing night DNs against each Night Service Group and Option.

The designated SCL consists of 100 two-digit translations. The first digit represents the Night Service Option number, while the second digit represents the Night Service Group number. Night Service Option zero (0) and Group zero (0) are reserved for the customer Night number, and therefore should not be defined, (i.e., 00, 01, 02, 03, 04, 05, 06, 07, 08, 09, 10, 20, 30, 40, 50, 60, 70, 80, and 90). The following is a sample Night Number Table with an explanation of how calls are terminated:

Option	Group	Number
.	.	
.	.	
.	.	
2	5	4311
2	6	4011
2	7	3893
.	.	
.	.	
3	5	3400
3	6	4321
3	7	4780
.	.	
.	.	

Night stations 4311, 4011, 3893 are assigned to Night Service Option 2 for Night Service Groups 5, 6, and 7 respectively.

If Night Service Option 2 is active, night calls from trunks designated in LD 14 as Night Service Group 5 will be routed to 4311, night calls from trunks designated in LD 14 as Night Service Group 6 will be routed to 4011, and night calls from trunks designated in LD 14 as Night Service Group 7 will be routed to 3893.

If the attendant selects Night Service Option 3, night calls from trunks designated in LD 14 as Night Service Group 5 will be routed to 3400, night calls from trunks designated in LD 14 as Night Service Group 6 will be routed to 4321, and night calls from trunks designated in LD 14 as Night Service Group 7 will be routed to 4780.

Attendant Console

This section describes the sequences to be followed by the attendant to select and query the Night Service Option and to activate Enhanced Night Service.

	ACTION	RESPONSE
1	Press Shift key	
2	Press Loop key	Indicator is activated.
3	Press Night key	Indicator flashes. Dial tone is received. Current Night Service Option number is displayed.
4	A) QUERY ONLY	
	i. Press RLS key	Indicator next to Loop and Night keys deactivates. Display is cleared.
	OR	
	B) SELECT	
	i. Dial a one-digit (0-9) option number.	Dial tone is removed. Old Night Service Option number (X) is shifted, new Option number (Y) is displayed, and X and Y are separated by a hyphen, (e.g., Y-X).
	ii. Press RLS key	Indicator next to Night and Position Busy keys deactivates. Night Service Option is stored. Display is cleared.
5	ACTIVATE Enhanced Night Service	
	Press Night key or Position Busy key if you are last active Attendant.	Indicators next to Night and Position Busy keys are activated. Current (active) Night Service Option number is displayed.

Introduced in X11 Release:	All
Networking:	No

No Hold Conference

Combined with Conference, Speed Call, System Speed Call, Autodial, and Hot Line, No Hold Conference (NHC) allows you to establish a Conference call without placing the current caller on hold.

This feature is available in four forms, merging No Hold Conference (NHC) with Autodial, Speed Call, and Hot Line into a single key. The new combined keys are the Conference-Autodial (CA), Conference-Speed Call (CS), and Conference-Hot Line (CH) feature keys. A No Hold Conference (NHC) key can also be configured, acting as a simple Conference key.

Conference-Hot Line can be used in the following two ways:

— The Direct CH option has the number stored with the key.

— The List CH option has a pointer that selects an entry from a Hot Line list.

When a telephone is connected to another party, you can originate a Conference-Autodial (CA), Conference-Speed Call (CS), or Conference-Hot Line (CH) call by pressing the CA, CS, CH, or NHC key. The system determines the destination as if it were a regular Autodial, Speed Call, or Hot Line call. The parties are conferenced in without holding. For example, a call comes in to the customer notifying the customer of a fire. The user wishes to notify the fire department of the emergency without placing the original caller on hold, and the number is stored on the Conference-Autodial key. By pressing the CA key, the customer establishes a Conference call. The fire department is notified and the original connection is maintained.

When you press the feature key, one of the following occurs:

— If the destination is an idle internal Directory Number (DN), that DN rings and the CA, CS, CH, or NHC lamp flashes (60 ipm). You hear no ringback tone.

— If the destination is a trunk with answer supervision, the trunk is seized and the key lamp flashes. The voice path is not established until an answer signal is received.

— When the destination is a trunk without answer supervision, the trunk is seized, the voice path is established, and the key lamp flashes. All tone signals provided by the far end (e.g., ringback) are heard by all parties involved in the Conference call. Calls on trunks without answer supervision are treated as answered after digit outpulsing is completed.

— When the intended destination is a busy internal DN, trunk, or route, the key lamp fast flashes (120 ipm). Press the active call key to cancel the attempt. The active call key is the key on which the call is established. It can be any key on which a regular Conference call can be made, including the DN key, Call Waiting, and Automatic Call Distribution (ACD) Incalls keys.

— In the case of network blocking, or if a conference port is unavailable, the key lamp fast flashes. Press the active call key to cancel the attempt.

— When the destination is an invalid entry (e.g., a vacant number, or an illegal list entry) the key lamp fast flashes. Press the active call key to cancel the attempt.

Pressing the active call key at any time before the called party responds cancels the attempt, returning the telephone to the state prior to pressing the CA, CS, CH, or NHC key.

If the call is answered, the key lamp goes off, and the called party is added to the existing conversation. By pressing the active call key, the last added party is released. These operations can be repeated as often as necessary, according to your network configuration, to add new parties to an existing conversation.

If the CA, CS, or CH keys are pressed at any time other than during a Conference call, they operate as a regular Autodial, Speed Call, or Hot Line key. Pressing the NHC key allows the user to dial the number desired for the Conference call.

Operating parameters

Assignable keys are limited to the number of keys available on your telephone.

NHC is available on Meridian 1 proprietary telephones with the CA, CS, CH, and NHC keys. It is not available on the M3000, analog (500/2500 type) telephones, or Attendant Consoles.

The Release (RLS) key has no effect while the key lamps are flashing or fast flashing. Other than during these stages, it can be used to abort the Conference call.

The CA key, like the regular Autodial key, is programmable from the telephone.

The CS and CH keys must have the Speed Call and Hot Line numbers assigned in LD 18.

Data calls are not supported.

All four keys can coexist with each other as well as with other Conference, Autodial, Speed Call, and Hot Line features.

Feature interactions

Automatic Redial

When an Automatic Redial (ARDL) call is not accepted by the calling party, the No Hold Conference (NHC) key is ignored.

Call Page Network Wide

A station set or Attendant Console that no hold conferences an external Call Page Network Wide (PAGENET) uncontrolled call is not blocked. However, an external PAGENET controlled call is blocked.

Centralized Attendant Services

Centralized Attendant Service (CAS) attendants are not supported.

Conference - Six Party

This feature can be enabled at any time that a regular Conference-6 feature can be activated.

Display of Calling Party Denied

Display information on sets involved in a No Hold Conference call is based on the individual Class of Service of each set.

Hot Line

The CH key supports only one-way Hot Line calls.

Meridian 911

In a Meridian 911 environmental, No Hold Conference calls are treated as internal calls and are linked to the low priority queue of the ACD DN.

Meridian 911 Call Abandon

M911 abandoned calls cannot be No Hold conferenced.

Recorded Announcement Trunk

A Recorded Announcement (RAN) Trunk cannot No Hold conferenced.

Off-Hook Alarm Security

Off-Hook Alarm Security treatment occurs when a telephone with ASCA Class of Service attempts an NHC call and the ASTM expires. The OHAS DN is conferenced in with the other conferees.

System Speed Call list

Whenever the CS key is programmed for a System Speed Call list, all calls made with that key are System Speed Calls.

Feature packaging

No Hold Conference capability is available when the following features are equipped:

— Autodial (ADL) for CA key configuration

— Speed Call User (SCU) if the CS key is configured

— Enhanced Hot Line (HOT) package 70 for the CH key, and

— System Speed Call (SSC) package 34 to configure CS or CH keys.

Feature implementation

LD 18 – Provision Speed Call or Hot Line numbers for CS and CH keys.

Prompt	Response	Description
REQ	NEW CHG	Add, or change a Speed Call list.
TYPE	SCL SSC HTL	Speed Call, System Speed Call, Hot Line.
CUST	0-99 0-31	Customer number (when TYPE = HTL). For Option 11C.
LNSO	0-8190	Speed Call list number.
NCOS	(0)-99	NCOS (when TYPE = SSC or HTL).
DNSZ	xx	Maximum number of digits in a list entry, where: xx = 4, 8, 12, (16), 20, 24, 28, or 31.
SIZE	1-1000	Maximum number of entries in the Speed Call list.
WRT	(YES) NO	Data is correct and list can be updated.
STOR	xxx	xxx = list entry number (0-9, 00-99, or 000-999).
	yy...yy	yy = digits to be stored against the entry (must be equal to or less than DNSZ).
WRT	NO (YES)	Data is correct and list can be updated.

Note: The WRT prompt follows the SIZE and STOR prompts asking you to confirm the correctness of the data just entered. If data is correct, enter YES or <CR>. A response of NO after the SIZE prompt causes all data entered to be ignored. A response of NO after the STOR prompt generates a warning message (SCH3213) indicating that the data was not stored and must be reentered.

A response of (*), aborts the program. Only the last STOR value is lost. All previous values to which WRT was YES are saved.

In X11 Release 17 and later, the following information is displayed with the WRT prompt, following SIZE: ADDS: MEM: xxxxx DISK: yy.y

Where xxxxx is the amount of protected memory and yy.y is the number of disk records required for the new speed call list. Check the MEM AVAIL and DISK REC AVAIL values displayed before the REQ prompt.

LD 11 – Add or change No Hold Conference for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx CA 4-(16)-23 y...y	Combined NHC and Autodial key, where: xx = key number, and y...y = target number stored in the key (maximum 23 digits).
	xx CH D yy z...z	Combined NHC and Direct Hot Line key, where: xx = key number yy = number of digits in the target number, and z...z = target number stored within the key.
	xx CH L 0-999	Combined NHC and Hot Line key, where: xx = key number, and 0-999 = Hot Line list entry.
	xx CS yyy	Combined NHC and Speed Call key, where: xx = key number, and yyy = Speed Call list number.
	xx NHC	NHC key, where: xx = key number.

Feature operation

No Hold Conference (NHC)

To establish an NHC call using the NHC key:

1 Establish a call.

2 Press **NHC**. The indicator goes on steadily.

3 Dial the number for the conference. The indicator flashes until the call is answered.

4 The conference is complete.

Conference-Autodial (CA)

To store an Autodial number:

1 Press **CA** (Conference-Autodial). The CA indicator flashes.

2 Enter the number.

3 Press **CA**. The indicator goes off.

To use Conference-Autodial:

1 Establish a call.

2 Press **CA**. The indicator flashes until the call is answered.

3 The conference is complete.

Conference-Hot Line (CH)

To establish an NHC call using the CH key:

1 Establish a call.

2 Press **CH** (Conference-Hot Line). The indicator flashes until the call is answered.

3 The conference is complete.

Conference-Speed Call (CS)

To establish an NHC call using the CS key:

1. Establish a call.
2. Press **CS** (Conference-Speed Call). The indicator goes on steadily.
3. Enter the Speed Call list entry number for the conference number. The indicator flashes until the call is answered.
4. The conference is complete.

Note: To disconnect the last NHC conference caller in any of the above procedures, press the DN key once.

Introduced in X11 Release:	19
Networking:	No

North American Numbering Plan

The North American Numbering Plan (NANP), established in 1947 and currently administered by Bellcore, governs the telephone numbering system throughout Bermuda, Canada, the Caribbean, and the United States.

Two components of the NANP are Interchangeable Numbering Plan Areas (INPAs) and Carrier Access Codes (CACs). NPAs are the three-digit prefixes commonly known as area codes. CACs permit telephone users to access any interexchange carrier or operator service provider. CACs must be supported by any entity, such as a hotel, motel, hospital, university, airport, gas station, or pay telephone owner, that makes telephone services available to the public.

Interchangeable Numbering Plan Area

The Interchangeable NPA codes plan was developed in the 1960s to manage the inevitable depletion of available codes. Prior to 1995, all area codes had an N(0/1)X format, where N was any digit from 2 to 9 inclusive and X was any digit, 0 to 9. As of January 1995, area codes have an NXX format, increasing the available codes to 640.

Modifications to X11 software, including changes to LDs that accept NPA or Home NPA codes, have eliminated dependencies and limitations associated with the old NPA code format.

The introduction of Interchangeable NPAs means that an area code (NPA) can appear identical to a Central Office prefix or a private network Location Code (LOC).

It is important to avoid conflicts among NPAs, Central Office prefixes, and LOCs. It is recommended that customers implement 1+ dialing to eliminate ambiguity.

Customers who use Autodial, Speed Call, or the Hot Line feature may need to modify the lists and tables associated with these features to accommodate the new prefixes or to reflect changes to numbers resulting from implementation of 1+ dialing.

The remainder of this section discusses the procedure that Basic Alternate Route Selection (BARS)/Network Alternate Route Selection (NARS) customers need to follow to handle the NPA changes. Although Alternate Route Selection (ARS) and Direct Trunk Access customers need not modify their databases, those who use Call Detail Recording and/or Toll Denied Class of Service should consider the effect of NPA changes on their operations.

BARS/NARS

BARS/NARS prohibits the entry of identical NPAs, Central Office prefixes, or LOCs. Typically, customers construct translation tables with NPA and LOC codes associated with one Access Code and Central Office codes associated with a second Access Code. Now that LOC and NPA codes may be identical, this option no longer guarantees that codes will not conflict.

Table 116 summarizes the options.

Table 116
Access Codes and 1+ dialing

# of Access Codes	Need LOC?	Use 1+?	Results
2	yes	yes	no conflicts
2	yes	no	may need to check that no LOC is identical to any NPA (depends on access code arrangement)
2	no	yes	no conflict
1	no	yes	no conflict
1	no	no	not recommended
1	yes	yes	not recommended

The ideal dialing plan continues to use two Access Codes, with 1+ dialing for NPA calls. (Digit Manipulation can remove the "1" for customers whose Central Office does not support 1+ dialing.)

Customers with two Access Codes that do not want to use 1+ dialing must ensure that no LOCS in the database are identical to existing NPAs. The database needs to be checked whenever a new NPA is introduced.

Customers who do not need LOCs can use a single Access Code and 1+ dialing or two Access Codes, one for NPA and one for the Central Office code.

Software modifications enable users to enter the new interchangeable NPAs in the following tables:

— Customer Data Block, LD 15. Changes allow the interchangeable NPA entry.

— Electronic Switched Networking (ESN) Translation tables, LD 90. Changes allow the interchangeable NPA format to be entered in response to NPA and HNPA prompts. Responses are compared to NARS/BARS call digits to determine call routing.

— Free Calling Area Screening (FCAS) tables, LD 87. Changes allow users to enter the interchangeable NPA format in response to the NPA prompt. Prompt values are compared to dial digits to determine if FCAS should screen calls.

— Feature Group D (FGD) Code Restriction tables, LD 19. Changes allow entry of the interchangeable NPA format in response to the NPA prompt. Feature Group D uses the response to restrict certain calls that terminate at, or tandem through, a given node.

— Meridian 911 (M911) Numbering Plan Digit/Information Digit (NPID) tables, LD 16. Changes allow entry of the interchangeable NPA format.

This software is available beginning with X11 Release 19. Upgrades may also require hardware modification depending on route selection capabilities, system type, and software release.

Direct Trunk Access and Alternate Route Selection

Direct Trunk Access and Alternate Route Selection customers need not update software to support interchangeable NPAs. Customers using Direct Trunk Access should continue to monitor local dialing procedures to ensure correct toll call recognition.

System upgrades

Upgrade requirements can include hardware and software. For specific information, consult *Upgrade system installation* (553-3001-250).

Feature implementation

The following prompts have been modified to accept NPA input in the new interchangeable format:

LD 15 – Home Numbering Plan Area modification.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB NET	Customer Data Block. Release 21 gate opener.
...		
- ISDN	YES	Change ISDN options.
- HNPA	200-999 1200-1999	Home Numbering Plan Area code.

LD 16 – NPA code definition for the M911 feature.

Prompt	Response	Description
...		
TYPE	NPID	Numbering Plan Digit/Information Digit table.
IDTB	0-7	NPID table number.
NPID	0-9	NPID to be translated.
TRMT	NPA	NPID treatment.
NPA	200-999	Numbering Plan Area code.

LD 19 – NPA input for incoming Feature Group D ANI screening.

Prompt	Response	Description
...		
TYPE	ANI	Feature Group D data block.
ANIT	(OVF), RAN xxx, DN xxx, NCOS xxx	Invalid Automatic Number Identification (ANI) treatment.
NPA	200-999	Three ANI digits in NPA format (prompt accepts only three digits even if 1+ dialing is in effect).

LD 87 – Free Call Area Screening definition.

Prompt	Response	Description
...		
FCI	xxx	Free Call Area Screening table index number.
NPA	200-999 200-999 200-999	Area code or extended NPA code translation (only three digits accepted even if 1+ dialing is in effect).

LD 90 – NARS/BARS.

Prompt	Response	Description
...		
TRAN	AC1, AC2, SUM	Access code 1, 2, or summary tables.
NPA	200-999 200-999 200-999 1200-1999 1200-1999 1200-1999	Area code or extended NPA code translation.
HNPA	200-999 1200-1999	Home Numbering Plan Area code.

Carrier Access Codes

A Carrier Access Code (CAC) gives a caller access to any interexchange carrier or Operator Service Provider (OSP). FCC regulations require that Call Aggregators, such as hotels, motels, hospitals, universities, airports, gas stations, and pay telephone owners, provide selective access to the public. Callers dial the CAC to reach their desired carrier or OSP before dialing the telephone number.

Aggregators, although they must allow callers access to any long distance caller, are permitted to block calls selectively. Selective equal access lets aggregators choose to block direct-dialed calls that result in charges to the originating telephone. Aggregators cannot block operator-assisted calls.

Northern Telecom provided an up-issue of X11 Release 14 in 1992 to conform to Federal Communications Commission (FCC) Equal Access requirements. Beginning with X11 Release 17, all software releases support Equal Access. X11 Releases 15 and 16 do not support Equal Access. Support for expanded codes, as described in the following paragraph, is available beginning with X11 Release 19.

The CAC has included a "10" identifying prefix followed by a three-digit Carrier Identification Code (CIC) for a total of five digits. New FCC regulations, reflected in X11 Release 19, require that the CAC expand to seven digits: a "101" identifying prefix followed by a four-digit CIC. The regulations require that both the old five-digit format and the new seven-digit format be supported for an 18-month permissive period during 1995 and 1996. After this period, only the longer format will be supported.

Feature packaging

Equal Access compliance is included in base X11 system software. The Network Class of Service package (NCOS) package 32 is required to configure Equal Access.

Feature implementation

Current Equal Access users who install new software prior to the end of the 1995/1996 FCC interim period must set the Original Carrier Access Code (OCAC) flag in LD 17 to YES when they upgrade their software or begin using a release that supports the CAC expansion feature.

For complete information on implementation and configuration, refer to the Equal Access Compliance feature description in this document.

Feature operation

X11 software allows the following operator-assisted North American and international dialing sequences:

— CAC + 0

— CAC + 0 + (NPA) + NXX + XXXX

— CAC + 01 + CC + NN

X11 software allows or denies these direct-dialed calls:

— CAC + 1 + (NPA) + NXX + XXXX

— CAC + 011 + CC + NN

where

CAC = Carrier Access Code (10XXX or 101XXXX)

NPA = Numbering Plan Area (area code)

NXX = Central Office code format
(N = any digit except 0 or 1; X = any digit 0–9)

XXXX = any four digits

CC = Country Code, and

NN = National number.

Introduced in X11 Release:	All
Networking:	No

Off-Hook Alarm Security

With X11 Release 18 and later, Off-Hook Alarm Security (OHAS) allows locked out calls to be intercepted to a customer-defined Directory Number (DN) other than an attendant (for example, a security DN). OHAS treatment is determined on a telephone basis by assigning a Class of Service called Alarm Security Allowed (ASCA). By enhancing line lockout, telephones with Alarm Security Allowed (ASCA) Class of Service are intercepted to customer-defined Directory Numbers (DNs) when the dial tone/interdigit timer expires or the telephone is Forced Out of Service (FSVC). Telephones without ASCA continue to use the existing line lockout treatment; refer to the Line Lockout module in this document.

An Off-Hook Alarm Security (OHAS) DN can be a Single Appearance Directory Number (DN), a Multiple Appearance DN, or an Automatic Call Distribution (ACD) DN. The OHAS DN cannot be an attendant DN, Listed DN, SPRE, Virtual ACD Agent, or Trunk Access Code.

To associate a telephone with an OHAS DN:

— Associate the OHID and FSVC (if necessary), and the Alarm Security Timer (ASTM) to an OHAS DN through the ODNx prompt in LD 15.

— Assign the ASCA Class of Service in LD 10 or LD 11.

If the ASCA Class of Service is assigned, but the telephone is not associated to an OHAS DN, an error message appears on the maintenance TTY when the system tries to redirect the call.

The Alarm Security Timer (ASTM) provides dial tone and interdigit timing for telephones with ASCA Class of Service. The ASTM does not apply to telephones being Forced Out of Service (FSVC).

Dial tone and interdigit timeout – call treatment

A telephone associated with an OHAS DN that receives a dial tone or interdigit timeout intercepts to the OHAS DN specified by the telephone's Off-Hook Interdigit OHAS number (OHID).

Forced Out of Service (FSVC) – call treatment

A digital telephone is considered FSVC when the line is cut, damaged, or unplugged.

The FSVC OHAS treatment applies only to digital telephones. A telephone associated with an OHAS DN that is FSVC intercepts to the OHAS DN specified by the telephone's FSVC number.

Multiple OHAS DNs

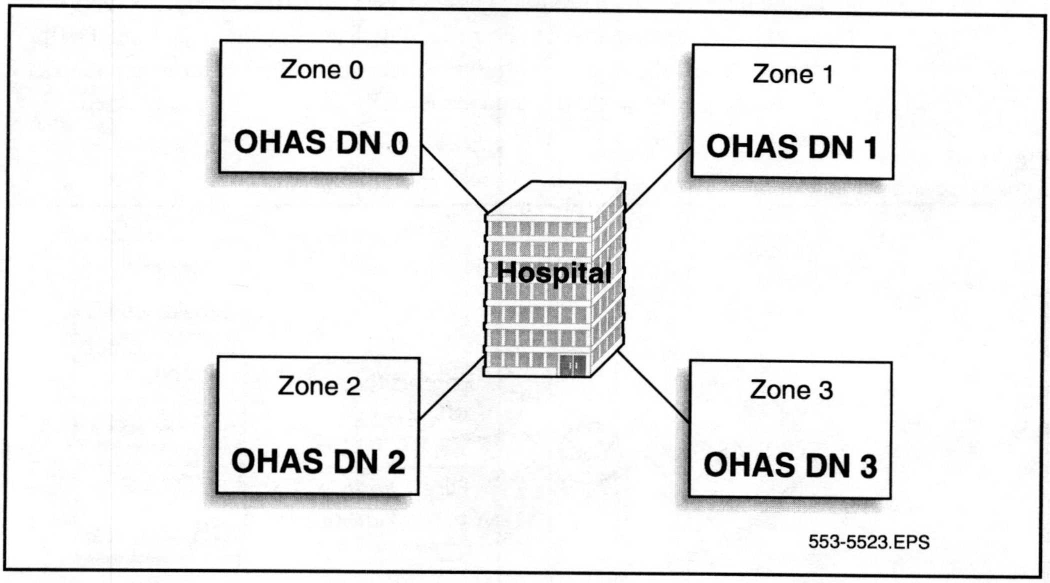
The two methods for handling multiple OHAS DNs are zone and event dependent, and are described in the following sections.

Multiple OHAS DNs – zone dependent

OHAS allows for multiple OHAS DNs within a single customer group, enabling the customer to create multiple zones.

For example, a hospital with several locations can define separate OHAS DNs for each location and define each distinct location as a zone. In Figure 68, the hospital has four zones. A separate OHAS DN is defined for each of the four zones. Zone 0 uses OHAS DN 0, Zone 1 uses OHAS DN 1, and so on. Each telephone in Zone 0 defines the OHID and FSVC numbers to 0; each telephone in Zone 1 defines the OHID and FSVC numbers to 1, and so on.

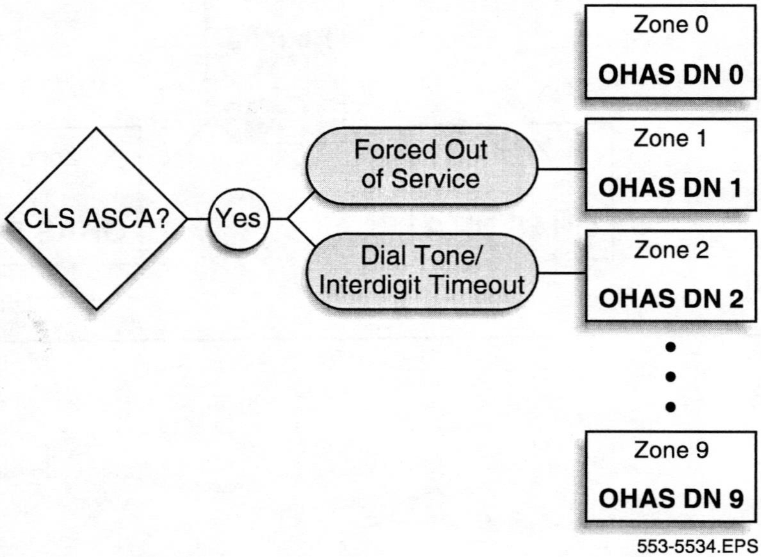
Figure 68
Zone dependent example



Multiple OHAS DNs – event dependent

OHAS can distinguish between OHID timeout and FSVC events by having a single telephone with separate OHAS DNs for OHID timeout and FSVC events (e.g., a telephone can be defined with a FSVC number 1 and OHID number 2. If a dial tone/interdigit timeout occurs, the telephone intercepts to OHAS DN 2. If the same telephone is FSVC, OHAS DN 1 is notified).

Figure 69
Event dependent example



OHAS TTY display

Every time an OHAS intercept treatment takes place, a message is sent to all maintenance TTYs. This message contains an OHAS message indicator, the originating DN and TN, and a time stamp.

Format			
OHASxxxx	<dn>	l s c u	time stamp
Output example			
OHAS0000	5003	1 0 1 0	04:30:21

Note: The two possible OHAS messages are:
OHAS0000OHAS treatment due to dial tone/interdigit timeout, and
OHAS0001OHAS treatment due to Forced Out of Service call treatment.

Operating parameters

OHAS is not supported for attendants or networks.

OHAS intercept treatment for FSVC telephones is provided only for the following telephones:

— the M2009, M2112, and M2018

— the M2317

— the M3000, and

— the M2006, M2216, M2616, M2008, and M2016.

The Alarm Security Timer (ASTM) does not apply to telephones being FSVC.

The timing for recognizing a FSVC condition depends on the type of card that the system is using:

— The Integrated Services Digital Line Cards (ISDLCs) take approximately six seconds to recognize an FSVC condition.

— Peripheral Controller cards take approximately one second to recognize an FSCV condition.

Once a trunk is seized, OHAS treatment does not apply.

Feature interactions

Call Redirection

Call Redirection features defined for telephones with ASCA Class of Service work as currently defined in the system. The Call Redirection features include the following:

- Call Forward All Calls
- Call Forward No Answer
- Call Forward Busy
- Call Forward by Call Type
- Call Pickup, and
- Hunting.

Call Transfer

A telephone receives the OHAS treatment if the telephone has ASCA Class of Service and attempts to transfer a call and the ASTM expires.

China – Flexible Feature Codes - Busy Number Redial

Busy Number Redial cannot be used on a set with Off-Hook Alarm Security Allowed, since ADL cannot be configured on these sets.

Conference

The OHAS line lockout treatment occurs when a telephone associated with an OHAS DN initiates a Conference call and the ASTM expires. Only the Conference initiator receives the OHAS treatment; other conferees remain in Conference. If the initiator of the Conference call presses the Conference key, the OHAS DN is conferenced in with the other conferees.

Electronic Switched Network
Trunk Access Codes

If an Electronic Switched Network or Trunk Access Code is dialed, the dial tone/interdigit timer is stopped and the telephone will not recall to the designated OHAS DN after the specified time period has elapsed.

Last Number Redial
Stored Number Redial

OHAS treatment may apply to these features if the ASTM expires.

Line Lockout

OHAS treatment occurs when a telephone with ASCA Class of Service receives an interdigit or dial tone timeout. The ASTM is used instead of the dial tone and interdigit timers (DIDT and DIND, respectively) normally used for LLT and DLT line lockout treatment.

Multi-Party Operations

Three-party Service (TSA) and Alarm Security Allowed (ASCA) Classes of Service are mutually exclusive. A set assigned TSA Class of Service cannot also be assigned ASCA Class of Service, and vice versa; a set assigned ASCA Class of Service cannot also be assigned TSA Class of Service.

No Hold Conference

OHAS treatment occurs when a telephone with ASCA Class of Service attempts an No Hold Conference call and the ASTM expires. The OHAS DN is conferenced in with the other conferees.

Room Status

OHAS takes precedence over the off-hook detection method of the Room Status feature. If a telephone is defined with the Alarm Security Allowed (ASCA) Class of Service, the off-hook detection method does not work.

Speed Call
Speed Call, System

OHAS treatment may apply to these features if the ASTM expires. The Alarm Security Timer may expire for the following reasons:

— A dial tone or interdigit timeout occurs while dialing the speed call access code.

— The Speed Call being accessed has an asterisk (*) causing a three-second delay. If the ASTM is three seconds or less, the OHAS intercept treatment may occur.

Feature packaging

Off-Hook Alarm Security is included in base X11 system software.

Feature implementation

LD 15 – Define the Off-Hook Alarm Security (OHAS) Directory Numbers (DNs). OHAS DNs must have ASCA Class of Service assigned in LD 10 or LD 11.

Prompt	Response	Description
REQ	NEW CHG	Add or change a customer.
TYPE	CDB INT	Customer Data Block. Release 21 gate opener.
CUST	0-99 0-31	Customer number. For Option 11.
- LLT	(OVF) ATN OFA	Flexible line lockout treatment.
...		
TYPE	OAS	Release 21 gate opener.
OHAS	(NO) YES	(Do not) change OHAS parameters. OHAS is prompted with X11 Release 20 and earlier.
The following prompts occur only if OHAS = YES, or if the Release 21 gate opener has been entered.		
- ODN0	xxx...x	OHAS DN 0.
- ODN1	xxx...x	OHAS DN 1.
- ODN2	xxx...x	OHAS DN 2.
- ODN3	xxx...x	OHAS DN 3.
- ODN4	xxx...x	OHAS DN 4.
- ODN5	xxx...x	OHAS DN 5.
- ODN6	xxx...x	OHAS DN 6.
- ODN7	xxx...x	OHAS DN 7.
- ODN8	xxx...x	OHAS DN 8.

- ODN9	xxx...x	OHAS DN 9.
- ASTM	1-(30)-63	The timer applies to all OHAS DNs and is programmable in one-second increments.

LD 10 – Assign Alarm Security Allowed (ASCA) Class of Service.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	500	Telephone type.
CLS	(ASCD) ASCA	Alarm Security (denied) allowed. When ASCA is assigned, the OHAS DN must be defined in LD 15.
OHID	(0)-9	Off-Hook Interdigit OHAS number.

LD 11 – Assign Alarm Security Allowed (ASCA) Class of Service.

Prompt	Response	Description
REQ	NEW CHG	Add or change.
TYPE	aaaa	Telephone type, where: aaaa = 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
CLS	(ASCD) ASCA	Alarm Security (denied) allowed. When ASCA is assigned, the OHAS DN must be defined in LD 15.
OHID	(0)-9	Off-Hook Interdigit OHAS number.
FSVC	(0)-9	FSVC OHAS DN number. The FSVC prompt is given only to digital telephones.

Feature operation

No specific operating procedures are required to use this feature.

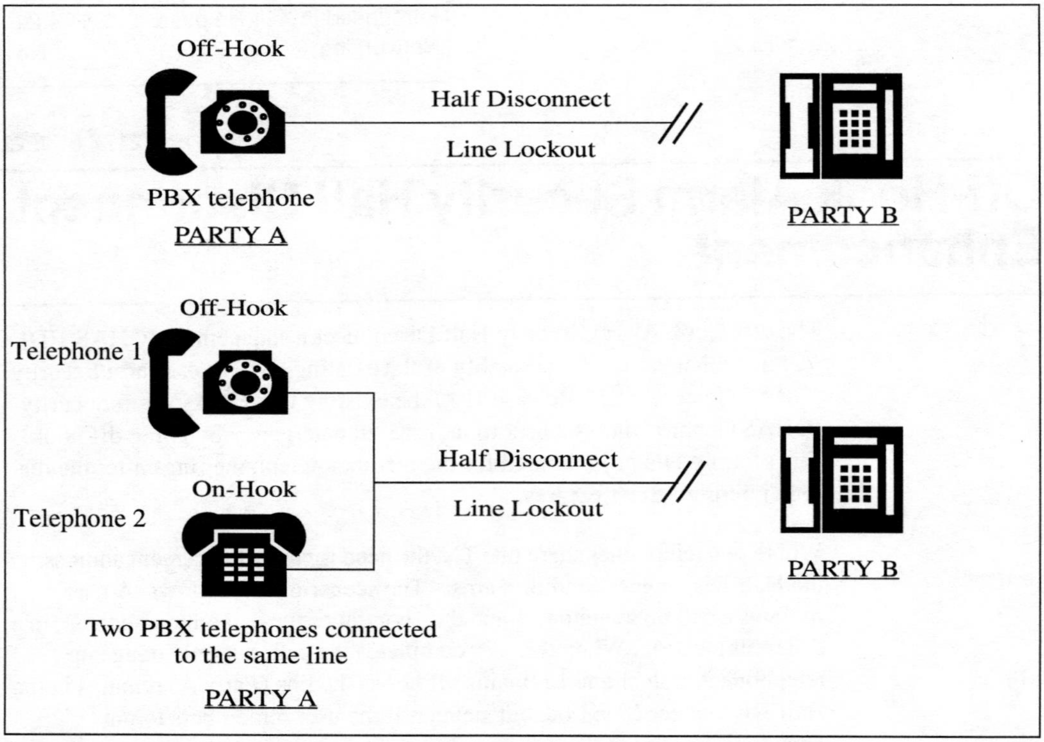
Introduced in X11 Release:	24.2x
Networking:	No

Off-Hook Alarm Security Half Disconnect Enhancement

The Off-Hook Alarm Security Half Disconnect Enhancement (OHAS HD) feature, enhances the functionality of the existing Off-Hook Alarm Security (OHAS) feature, (X11 Release 18). The existing Off-Hook Alarm Security (OHAS) feature allows a user to indicate an emergency by going off hook. The security DN programmed for the off-hook telephone rings after the dial tone/interdigit timer expires.

Where two telephones share one TN, the need for an enhancement addressing the Half Disconnect condition arose. The scenario is as follows. A user initiates a call on telephone 1 and then continues the call on telephone 2, in a different location. When the user completes the call, but only hangs up telephone 2, (telephone 1 remains off hook) the line (Party A) remains in the Half Disconnect/Line Lockout state until the user remembers to put telephone 1 on hook. (See Figure 70.)

Figure 70
OHAS HD Scenario



When the OHAS HD feature is enabled you can define, on a customer basis, the length of time before the OHAS HD treatment is given. When this timer expires the programmed security DN rings. If a telephone goes on-hook before the OHAS Half Disconnect Timer (HDTM) expires, the OHAS Half Disconnect treatment is canceled, as the telephone has completed its disconnect.

The OHAS Half Disconnect Option, (HDOPT) determines the number of OHAS Half Disconnect treatments that can be given to telephones that remain in the Half Disconnect state. This is programmed on a customer group basis.

There are three OHAS HD options for Half Disconnected telephones with OHAS enabled.

- **HDOPT = 0** is the existing treatment without the OHAS Half Disconnect Enhancement. It is the default option and disables the Off-Hook Alarm Security Half Disconnect feature. Line Lockout treatment occurs after the normal Half Disconnect timer expires and Half Disconnect state is recognized.
- **HDOPT = 1-10** indicates the maximum number of OHAS HD treatments given to the half disconnected analog (500/2500) type telephone. This option allows a limited number of OHAS HD treatments. If the telephone remains off-hook in the half-disconnect state after the maximum number of treatments has expired, Line Lockout occurs when the security DN disconnects after the last OHAS Half Disconnect treatment.
- **HDOPT = CONT** provides a continuous application of the OHAS HD treatment, while the analog telephone remains in the half disconnected state. This option continues to call the security DN every time the HDTM expires until the analog (500/2500) type telephone goes on-hook.

Class of Service

To enable the OHAS HD feature the telephone must have CLS = Alarm Security Allowed (ASCA). Therefore when the HDTM timer expires, instead of giving the Line Lockout treatment, the OHAS HD treatment is given.

OHAS security DN

On a telephone basis an HDID is assigned. The HDID is the OHAS HD Index number. The values are 0 - 9. The Index number refers to the ten OHAS DNs you can program in the Customer Data Block. For example, if a telephone has HDID 1 assigned, OHAS HD treatment calls the security DN programmed for OHAS DN 1 (ODN1), in the Customer Data Block.

The OHAS HD Index can be configured to send calls to the same security DN as the existing OHAS Off-Hook Index (OHID) or a different security DN. This flexibility allows you to distinguish between regular OHAS dial tone/interdigit time-out treatment calls (emergency situations) and OHAS HD treatments for half disconnect calls.

OHAS Half Disconnect Timer

With the OHAS Half Disconnect Enhancement feature enabled, the administrator can define the length of time before the OHAS HD treatment is given. The OHAS HD timer (HDTM) gives the average user enough time to complete the disconnect of the previous call by placing all the analog telephones on-hook. The length of the OHAS Half Disconnect timer can be defined from 1 to 600 seconds (10 minutes). The timer is programmable in one second increments. The HDTM starts after the half disconnect state is detected. The default for the HDTM is 30 seconds.

OHAS TTY record display

As with the existing OHAS feature, a message also prints out on the TTY terminal indicating the telephone which is receiving OHAS treatment. The message is the same for regular OHAS and OHAS Half Disconnect.

Each occurrence of an OHAS HD intercept treatment results in a message printout on the service change TTY or the active TTY. The content and the format of the OHAS HD message is the same as the regular OHAS off-hook or interdigit time-out message.

The following is an example of the record content:

```
OHAS000 2010 1 0 1 3 5:04:04  7/09/1998
```

The definitions of the fields are as follows:

OHAS000 = OHAS message indicator

2010 = DN (the DN of the analog (500/2500) type telephone receiving OHAS or OHAS Half Disconnect treatment)

1 0 1 3 = l s c u (the TN of the analog (500/2500) type telephone receiving OHAS or OHAS Half Disconnect Treatment)

> ***Note:*** The TN of the Option 11 is only two digits (c u).

5:04:04 = time stamp (when the OHAS or OHAS Half Disconnect Treatment is given)

7/09/1998 = date stamp

Operating parameters

While a 500/2500 telephone is in the half disconnect/Line Lockout state, the OHAS feature for emergencies cannot be triggered. OHAS will not work until the off-hook 500/2500 telephone goes on hook to disconnect the previous connection.

When OHAS Half Disconnect occurs, new calls cannot be initiated from the half-disconnected telephones.

If Party A goes on-hook at any time, the OHAS Half Disconnect treatment is canceled, since the disconnect is completed.

The OHAS Half Disconnect Timer is separate from the existing OHAS timer.

Digital telephones do not go into the half disconnect state. Digital telephones cannot share a TN with other telephones.

The feature does not apply to digital telephones since the half disconnect state does not apply to them.

The OHAS HD treatment is not provided for Attendant Consoles.

If the telephone remains off-hook in the half-disconnect state after the maximum number of OHAS HD treatments has expired, Line lockout occurs when the security DN disconnects after the last OHAS Half Disconnect treatment.

OHAS HD calls can be directed to a separate security DN to enable the user who answers the calls to distinguish between an Off Hook Alarm Security call and an Off Hook Alarm Security Half Disconnect Call.

Ringback tone can be heard at the off-hook analog telephone when the security DN is ringing. Anyone who uses one of the half-disconnected 500/2500 telephones can speak to the person who answers the security DN.

If Party A goes on-hook at any time, the OHAS Half Disconnect treatment is canceled, since the disconnect is completed.

If the connection is a trunk call and the far end does not disconnect completely, Party A will not go into the half disconnect state. The system treats Party B and Party A as if they are still on an active call.

The OHAS HD feature applies only to a single switch. It is not supported in a networking environment.

The OHAS HD security DN cannot be an Attendant DN.

The operation of the OHAS HD timer is impacted on systems with high traffic.

Feature interactions

Call Redirection

Call Redirection features defined for OHAS Half Disconnect security DN work as currently defined in the system. Call Redirection features include:

— Call Forward All Calls

— Call Forward No Answer

— Call Forward Busy

— Call Forward by Call Type

— Call Pickup

— Hunting

Conference

If an analog 500/2500 telephone user with the ASCA Class of Service is in a conference and all the other parties disconnect from the call while the user's telephone remains off hook, the OHAS Half Disconnect Enhancement feature applies to the half-disconnected telephone.

Line Lockout

If an analog telephone has the ASCA Class of Service, and it is in the half disconnected state, the OHAS HD treatment occurs if the customer-based OHAS Half disconnect option (HDOPT) is enabled. Choose HDOPT 1-10 or HDOPT = CONT. If HDOPT= 0 is selected, Line Lockout will occur.

If the telephone stays in the half disconnected state and the number of the OHAS HD treatments given to the telephone exceeds the maximum defined number, Line Lockout is given to the telephone after the last OHAS Half Disconnect treatment is given.

No Hold conference

The OHAS HD treatment works the same for a conference call initiated using No Hold Conference as for Conference.

Feature packaging

The Off-Hook Alarm Security Disconnect Enhancement is included in base X11 system software.

Feature implementation

This section contains the overlay procedures required to configure the Off-Hook Alarm Security Half Disconnect Enhancement feature on a Meridian 1.

The required procedures are as follows:

1. Configure OHAS Directory Numbers, the OHAS Half Disconnect treatment option and the OHAS HD timer in the Customer Data Block (LD 15).
2. Assign an ASCA Class of Service to the telephone (LD 10). Associate the telephone with one of the ten OHAS DNs configured in LD 15.

Note: The telephone is also programmed with an OHID, related to the OHAS feature.

LD 15 – Configure Off-Hook Alarm Security (OHAS) Directory Numbers (DNs), Half Disconnect treatment option, and the OHAS Half Disconnect timer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	OAS	Off-Hook Alarm Security (OHAS) options.
CUST	xx	Customer number.
ODN0	xxxx	OHAS DN 0.
...		
ODN9	xxxx	OHAS DN 9.

ASTM	1 - (30) - 63	OHAS off-hook or interdigit timeout timer in seconds.
HDOPT		OHAS Half Disconnect treatment options:
	(0) 1-10 CONT	No OHAS HD treatment given. Maximum number of OHAS HD treatments. Continuous OHAS HD treatments.
HDTM	1- (30) - 600	OHAS Half Disconnect timer in seconds (in increments of 1 second).

LD 10 - Assign an ASCA Class of Service to the telephone. Associate the telephone with one of the ten Off-Hook Alarm Security Directory Numbers (ODN0-9) configured in LD 15.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	500/2500 telephones.
TN		Terminal Number
	l s c u c u	Options 51-81C. Option 11C
CUST	xx	Customer Number.
DES	d..d	Office Data Administration System Station Designator.
. . .	. . .	
DN	x...x	Directory Number.
....		
CLS	ASCA	Alarm Security Allowed. (ASCD) = Alarm Security Denied is the default.
...		
OHID	(0) - 9	OHAS ID index to OHAS security DN.
HDID	(0) - 9	OHAS Half Disconnect Index to OHAS HD security DN.

Feature Operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	All
Networking:	No

Office Data Administration System

The Office Data Administration System (ODAS) package provides a method for retrieving administrative information stored in Meridian 1 memory. This feature can expedite administration and billing activities by significantly reducing the need for manual records.

The Station Line Designator (DES) code is any alphanumeric code of one to six characters. The customer selects this number, which can help the customer group telephones according to users, floor location, or any other category.

The following table lists the types of data that can be printed using the Office Data Administration System (ODAS) and the overlay program to use for each task:

Type of data required	LD
Count telephones with specified feature(s)	81
List Directory Number (DN) blocks by DATE entry	22
List DN blocks by station line designator (DES) entry	22
List Terminal Number (TN) alphabetically by DES	83
List TN with specified DATE entry	20
List TN with specified DES entry	20
Print Multiple Appearance Groups	82
Print TN to DES correlation for specified feature(s)	81
Print TN data blocks with specified DATE entry	20
Print TN data blocks with specified DES entry	20

Refer to the Northern Telecom Publication *Office Data Administration System description and engineering* (553-2721-100) for a complete description of the Office Data Administration System (ODAS).

Operating parameters

It is recommended that 1200 baud printers be used on larger systems to reduce the time required to obtain ODAS printouts. When a system is equipped with a 1200-baud printer, a 300-baud device must not be assigned to perform the same function.

Feature interactions

Refer to *Office Data Administration System description and engineering* (553-2721-100).

Feature packaging

Office Data Administration System (ODAS) package 20, has no feature package dependencies.

Feature implementation

LD 84/85 – Assign or change station line designator (DES) entry for telephones.

Prompt	Response	Description
TN	l s c u c u	Terminal Number. For Option 11C.
DES	a...x	DES (one to six alphanumeric characters).

LD 10 – Assign or change DES entry for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
DES	a...x	DES (one to six alphanumeric characters).

LD 11 – Assign or change DES entry for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
DES	a...x	DES (one to six alphanumeric characters).

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	All
Networking:	No

Off-Premise Extension

The Off-Premise Extension (OPX) feature allows a single line telephone serving as an extension to be located away from the customer premises. The loop limit is 1400 ohms to the station or equivalent long-line circuit interface. Distance varies depending on the gauge of wire used.

Refer to Northern Telecom Publication *500/2500 line cards description and operation* (553-2201-183) for additional information.

Operating parameters

The Off-Premise Extension (OPX) feature applies only to single line telephones. A QPC192 line circuit pack must be equipped.

Feature interactions

Refer to *500/2500 line cards description and operation* (553-2201-183).

Feature packaging

Off-Premise Extension (OPX) is included in base X11 system software.

Feature implementation

LD 10 – Add or change Off-Premise Extension Class of Service for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	(ONP) OPX	Telephone is an on-premises or off-premises extension.

Feature operation

There is no specific procedure required to operate this feature.

Off-Premise Station Analog Line Card

The Eight-port Off-Premise Station (XOPS) analog line card (NT1R20) is being introduced in Release 20 for North America and China. This development is part of the Global Line Card program.

The XOPS card supports the current portfolio of peripheral equipment, and is designed for use in Off-premises Station (OPS) environments, connected through a Central Office (CO)/Public Exchange. It is also suited for campus system environments. Each of the units on the card can be configured to be operated as an OPS extension or in an On-premises (ONS) configuration.

The new XOPS card requires a set of new downloadable parameters for Termination and Balance Impedance values. These parameters are downloaded to the card whenever it is initialized or enabled. In addition, the analog cards require the loss/levels to be set for each unit on the card using the B34 Flexible Level message interface. ONS units receive loss/levels statically on Initialize or Enable.

Operating parameters

The XOPS card requires a Main Distribution Frame (MDF) wiring installation plan similar to trunks, rather than other line cards. Therefore, it will not be possible to interchange the XOPS card with another line card without rewiring the connections, or adjusting the Terminal Numbers (TNs) using service change.

The Classes of Service have been renamed to be consistent with industry standard terminology as follows: OPX is now called OPS; and ONP is now called ONS.

New XOPS loss levels are also assumed for the EPE OPS units. Therefore, there will be a slight deviation in loss levels at an EPE OPS connection. Systems with both EPE OPS and XOPS are not recommended.

The jumper settings must be set in accordance with OPS and ONS Classes of Service.

No software support is provided for Answer Supervision in Release 20. The XOPS hardware will support Answer Supervision through Battery Reversal or Flash Hook.

No software support is provided for any Loopback from Extended Network Card (XNET) or XPEC to the XOPS line card.

The new XOPS line card uses B34 CODEC and Enhanced Extended Universal Trunk Card (EXUT) trunk circuitry. Therefore, the downloadable Termination Impedance (TIMP)/Balance Impedance (BIMP) combination parameter set, as defined for IPE EXUT, is likewise defined for the XOPS. The usage of TIMP/BIMP implies a limited number of downloadable combinations.

The XOPS is designed to work in North America using dynamic pad switching based on OPS and ONS Classes of Service. The card functions in a Static Loss Plan Download environment, but only the static levels associated with Analog Line Unit Short (ALUS) and Analog Line Unit Long (ALUL) are supported. In these situations, only Class of Service Long Line (LOL) or Short Line (SHL) has any meaning; OPS/ONS Class of Service of the unit is ignored.

As with the existing design, parameter download is not performed as part of enabling a Superloop, but is done as part of an initialization, or enabling of a unit, card, or peripheral shelf.

Hardware is compatible with the Meridian SL-100 PBX, but software support for the Meridian SL-100 is not included as part of the XOPS feature.

Feature interactions

Due to the Loss Planning requirements for the XOPS card, the Global Line Card feature interacts with other Loss Planning features. The XOPS card must be able to operate in system environments that are using North American Transmission Plan, Static Loss Plan Download (SLPD), or Dynamic Loss Switching (DLS).

Feature packaging

No new software packages have been introduced for this feature; however, Meridian 1 Superloop (XPE) package 203 is required, because the XOPS card can only operate in an IPE environment.

Feature implementation

In order to implement the new double density support for XOPS on a Superloop, the administrative programs require modification. In addition, download parameters can be installed or modified using service change.

LD 10 – The card density field is used to verify that unit and shelf fields in the external TN entered at the TN prompt are appropriate for the card on which it is configured. New checks allow an XOPS card to be configured as a Double Density card on a Superloop.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	500	Analog (500/2500 type) telephone data block. (Also 500M for the Option 11.)
TN	l s c u c u	Terminal Number. For Option 11C.
CDEN	SD DD 4D	Single, Double, or Quad Density.
DES	dddddd	1-6 alphanumeric character Office Data Administration System (ODAS) Station Designator.
...		

CLS	(OPS) (ONS) (LOL) (SHL)	Classes of Service ONS and OPS are supported. OPS is the default if the TN is on XOPS, otherwise ONS is the default. Classes of Service LOL and SHL are supported, but are not used for North America Loss Plan handling. LOL is the default if the TN is XOPS, otherwise SHL is the default.
...		
TIMP	(600) 900	Termination Impedance for XOPS unit. Prompted only if the specified TN is to be configured on an XOPS card (Double Density card on a Superloop).
BIMP	(3CM2) (600) 3COM, 900	Balance Impedance for XOPS unit. 3CM2 is the default if the CLS is OPS, otherwise the default is 600.

LD 10 – The commands for creating or modifying an analog (500/2500 type) telephone type logical card block are modified to support the new card density for the XOPS card.

Prompt	**Response**	**Description**
REQ	NEW CHG	New, or change.
TYPE	CARDSLT	Card block for single line terminations.
TN	l s c u c u	Terminal Number. For Option 11C.
CDEN	SD DD 4D	Single, double, or quad density.

LD 10 – The "Easy Change" option, used to change a single value associated with a unit, is supported. This command functions as if the expanded Change command were used to change only the BIMP and/or TIMP value, or to change the card density.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Analog (500/2500 type) telephone data block.
TN	l s c u c u	Terminal Number. For Option 11C.
ECHG	(NO) YES	(Deny) allow the Easy Change option.
ITEM	TIMP ttt, BIMP bbbb CDEN cc CLS sss sss	Prompted only if the response to ECHG is yes. New ITEM responses TIMP or BIMP have been added, with the associated responses for each item (ECHG of TIMP and BIMP are only allowable for a Double Density card on a Superloop (XOPS)). TIMP: ttt is 600 or 900. BIMP is prompted next. BIMP: bbbb is 3CM2, 600, 900 or 3COM (BIMP should be set to 600 if the unit is configured with ONS Class of Service). ITEM is prompted next. CDEN cc is SD, DD or 4D (ECHG of CDEN continues to be supported, but existing code ensures that a single density card with at least one unit with Class of Service OPS (an EPE OPS) cannot be changed to any other density. If CLS is changed to OPS, ONS, LOL, or SHL, TIMP is prompted next. Otherwise ITEM is prompted next.
TIMP	tttt	Prompted only if the response to ITEM was CLS of OPS, ONS, LOL, or SHL, and if CLS was changed from its previous setting. tttt is 600 or 900.
BIMP	bbbb	Prompted only if the response to ITEM is TIMP ttt or on change of CLS sss (bbbb is 3CM2, 600, 900, or 3COM).
ITEM	<CR>	Used to exit the ITEM prompt loop.

LD 10 – Additional checking is added to support MOV commands on XOPS units.

Prompt	Response	Description
REQ	MOV	Move.
TYPE	500	Analog (500/2500 type) telephone data block.
TN	l s c u c u	Existing TN. TN for the Option 11.
TOTN	l s c u c u	Destination TN. Destination TN for the Option 11.

LD 10 – Additional checks are added to support CPY (copy) commands involving XOPS units.

Prompt	Response	Description
REQ	CPY xx	Copy.
TYPE	500	Analog (500/2500 type) telephone data block.
...		
CFTN	l s c u c u	Copy from Terminal Number, prompted if REQ = CPY. For the Option 11.
SFMT	AUTO, DN, etc.	For AUTO and DN format types, the TNs are provided by the system.

LD 25 – Move card TNs from Superloop to Superloop.

Prompt	Response	Description
REQ	MOV	Move.
CUST	<CR>	Customer number.
SRCL	0-156	Source loop.
DSTL	0-156	Destination loop.
MVSG	(NO) YES	Move segment.
SCHD	l s c u TO l s c u cu TO cu	If attempting to move a Quad Density or Octal Density card on a Superloop to an XOPS card, or vice versa, an SCH6400 error message will be issued. For Option 11C.

LD 25 – Move card TNs from non-Superloop to Superloop.

Prompt	Response	Description
REQ	MOV	Move.
CUST	<CR>	Customer number.
SRCL	0-156	Source loop.
DSTL	0-156	Destination loop.
MVSG	(NO) YES	Move segment.
SCHD	l s c u TO l s c u c u TO cu	If attempting to move a Single Density, Double Density, or Quad Density card on a Superloop to an XOPS card, an SCH6400 error message will be issued. For Option 11C.

LD 97 – Install or customize Static Loss Plan Download table.

Prompt	Response	Description
REQ	CHG PRT	Change, or print.
TYPE	LOSP XCTP XPE SUPL XNPD SYSP	Install or change the system Loss Plan.
TTYP	STAT	Modify the system SLPD table.
NATP	YES NO	North American Transmission Plan.
STYP	PRED CSTM DISL	Static Loss Plan Download table type, where: PRED = Predefined table, CSTM = Customized table. DISL = Disable current active table If the response is PRED, TNUM is prompted. If CSTM is selected, SLPD port types are prompted after password verification. If response DISL is selected, SLPD will be disabled after password verification. If <CR> is entered, the table type is not changed (previously <CR> was treated as PRED).
TNUM	nn	SLDP Table number. nn is 1 to 25 Prompted if PRED is selected (REQ is prompted next).
PWD2	pppp ppp...p	Prompted only if STYP is CSTM and LAPW is restricted or the user logged in with the PWD1 password.
COTS	Rx Tx	CO trunk with SHL CLS.

LD 97 – This overlay is used to install or customize a Dynamic Loss Switching Alternate Levels table.

Prompt	Response	Description
REQ	CHG PRT	Change, or print.
TYPE	LOSP XCTP XPE SUPL XNPD SYSP	Install or change the system Loss Plan.
NATP	YES NO	North American Transmission Plan.
TTYP	DYNM	Modify the system DLS Alternate Levels table.
DTYP	PRED CSTM DISL	DLS Alternate Levels table type. If the response is PRED, TNUM is prompted. If CSTM is selected, DSL port types are prompted after password verification. If the response DISL is selected, DLS will be disabled after password verification. If <CR> is entered, the table type is not changed (previously <CR> was treated as PRED).
TNUM	nn	DLS Alternate Levels table number. nn is 1 to 3. Prompted if PRED is selected (REQ is prompted next).
PWD2	ppp ppp...p	Prompted only if DTYP is CSTM and LAPW is restricted or the user logged in with the PWD1 password.
COTS	Rx Tx	CO trunk with SHL CLS.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	20.0x
Networking:	No

On Hold on Loudspeaker

The On Hold on Loudspeaker (OHOL) feature is designed for brokers (dealers), and requires proprietary hardware to make use of its functionality. This feature provides brokers with the capability to monitor one (with the proprietary loudspeaker) or several (with the proprietary speech monitor) stock markets, while talking to one or several customers using the handset.

At any time the user can enter the call being monitored on the loudspeaker. This can also be done for the speech monitor unit either publicly by using the built in microphone (if provided) and the conversation will be heard on the channel, or privately by taking the call on the handset. Speech monitors work as loudspeakers, but with up to eight channels.

Operating parameters

This feature requires either proprietary loudspeakers that connect to M2616 sets, or a speech monitor system, and speech monitor units to work properly.

This feature is dependent upon access to conference cards and therefore each proprietary loudspeaker/speech monitor should have a conference loop assigned. Since the conference loops are used by the entire system, an option to separate normal conference traffic from "Dealer Group Traffic" is introduced. This is done by introducing a new response to the conference prompt in LD 17 making it possible to define a conference loop as a Dealer Conference loop. Conference loops designated as Dealer Conference loops are ordinary conference cards, but are used exclusively by the OHOL feature. The system manager is able to control how many parties can access a conference card and therefore decreases the risk of a blocking situation occurring.

Also, to avoid conference blocking enough conference loops designated as Dealer Conference have to be configured.

One conference loop per system can be assigned as a Spare Dealer Conference loop. This loop is used as a backup if the conference loop assigned to an OHOL unit is in invalid state. This loop can only be used by the OHOL feature.

Feature interactions

Attendant Barge-in
Attendant Break-in
Attendant Busy Verify
Override

It will not be possible to Break-in/Barge-in/Busy Verify/Override into a call on loudspeaker as it is effectively on hold at the set.

Audible Reminder of Held Call

This feature works with the OHOL feature as for normal calls on hold (that is, it gives a reminder there are calls on hold). Therefore, it is not recommended to use this feature with the OHOL feature.

Call Forward All Types

No type of call forward can be activated on a set with Speaker Allowed Class of Service.

Call Transfer
Conference

It will not be possible to transfer or conference the loudspeaker call to another party.

Call Waiting
Camp-on
Ring Again

These features can be applied to a busy loudspeaker DN.

Conference Loops

The configuration of conference loops has been modified to indicate whether a conference loop is a Dealer or an ordinary conference loop.

Dial Access to Group Call

If a group call is initiated from a set with Dealer Allowed (Class of Service), the conference is built up on the assigned loop of the loudspeaker or speech monitor system channel since this is a potential OHOL call.

Group Hunt

Group Hunt to a loudspeaker DN can be programmed, but will be ignored if configured as Make Set Busy (MSB) by call processing.

Group Hunt

Group Hunt to a loudspeaker DN can be programmed, but will be ignored if configured as Make Set Busy (MSB) by call processing.

Held Call Clearing

Going on-hook when Held Call Clearing is activated will clear the loudspeaker as for a normal held call. Therefore, it is recommended not to use this feature with the OHOL feature.

Hold

The feature is limited to use with normal hold or automatic hold. Deluxe hold will be ignored by call processing.

Hot Line
Voice Call

It is possible to program these keys with a loudspeaker DN, but operation will be the same as for direct dial to a loudspeaker DN.

Hot Line Two Way

This feature can be used with the speech monitor system. The DN of the speech monitor system channel is configured as the DN for the HOT line key.

Hunting
Call Forward

Hunt/Call Forward to a loudspeaker DN can be programmed, but will receive intercept treatment as for direct dial to the loudspeaker DN.

Music

If Music on Hold is equipped it will not be heard by either party during a loudspeaker call.

Ring Hold LED Status

This feature reverses the lamp indication of ringing and held calls. With this feature activated, held calls will fast flash and ringing calls will slow flash.

Single Call Ringing

If a single call ringing loudspeaker DN (a analog (500/2500 type) telephone with CLS = SPKA) is dialed, intercept treatment is provided.

Telephones - Analog (500/2500 type)

The loudspeaker and speech monitor system channels are configured as 500/2500 sets with Speaker Allowed Class of Service (CLS = SPKA). These sets are in a permanent off-hook state. The units are recognized as in lockout state by the system. These sets have to be assigned a conference loop (DCLP prompt in LD 10).

Feature packaging

On Hold on Loudspeaker (OHOL) package 196 is required to operate this feature.

It is recommended to have the Autohold feature configured with this feature to simplify its operation.

Feature implementation

LD 17 – Assign Dealer Conference loop and Spare Dealer Conference loop.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CEQU	Release 19 gate opener.
...		
- CONF	0-158	Conference loops.
	D0-D158	Conference loop number assigned as Dealer Conference loop.
	S0-S158	Conference loop assigned as Spare Dealer Conference loop. It is strongly recommended that this loop is in the same group as the unit planning to use this loop to minimize the use of intergroup timeslots.
	X0-X158	To remove entry.

LD 10 – A new Class of Service is added to this overlay to allow an analog (500/2500 type) telephone to be assigned as a loudspeaker DN.

A new prompt, DCLP (Dealer Conference Loop), has been added to configure the assigned conference loop.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	500	Analog (500/2500 type) telephone.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	SPKA	Speaker allowed.
DCLP	xx	Assign loop number with or without option Dealer Conference loop.

LD 11 – Configure the M2616 set with LSPK key. Only one key can be configured per set.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	M2616	Meridian Modular set.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	DELA	Dealer allowed.
KEY	xx LSPK nnnnnn	Loudspeaker, where xx is the key number, and nnnnnn is the LSPK DN which is the same DN as for the OHOL unit.

LD 11 – Configure a set with a DN key corresponding to a speech monitor system channel.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	M2616	Meridian Modular set.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	DELA	Dealer allowed.
KEY	xx SCR nnnnnn	xx is the key number. nnnnnn is DN which is the same DN as for the speech monitor system channel. When this DN is put on hold, the speech monitor unit will automatically be switched on.

Feature operation

Proprietary Loudspeaker System

This system consists of a M2616 set with a Loudspeaker (LSPK) key configured and an attached add-on module which has been modified to work as a loudspeaker. The proprietary loudspeaker is to be used when a user needs to be able to monitor one call on the loudspeaker at the same time as monitoring another call on the handset.

The loudspeaker is connected to a 500 line card and is in a permanent off-hook state. The DN of the loudspeaker must be Single Call Ringing (SCR).

Sets with this configuration are allowed to manually put calls onto the loudspeaker. The call to be put onto the loudspeaker has to be on hold at the set. To activate the loudspeaker, press the LSPK key and then press any DN key on hold. The held call is put onto the loudspeaker and will be heard publicly. A user can enter into the call by using the handset on the loudspeaker (if provided). While the loudspeaker is active, any other call will be maintained on the handset. More than one call can be put on hold on the set, however only one call at a time can be switched to the loudspeaker.

To release the call from the loudspeaker, the active call on the handset has to be put on hold (either by automatic hold or manual hold) or released.

Attempts to activate a call onto the loudspeaker when busy will be ignored.

Speech Monitor System

The speech monitor system is used in an environment where several users need to listen to the same call publicly. The speech monitor system enables calls to be automatically extended to a loudspeaker. The loudspeaker in this scenario is the speech monitor unit.

The speech monitor unit has a number of speech monitor system channels (a maximum of eight) available. These channels can be switched onto the speech monitor unit and heard publicly. Each speech monitor system channel has a SCR DN configured. This SCR DN has a mixed appearance on a key (DN or HOT) on a user's set. Several users can have the same mixed DN on their set (Multiple Appearance SCR DN). The set can also have a two-way HOT line key with the same DN as a speech monitor system channel. While monitoring up to eight calls on the speech monitor unit, the users' handsets are free to maintain other calls.

The speech monitor system channels are attached to a 500 line card which is in a permanently off-hook state. The unit is recognized as in lockout state by the system.

The speech monitor system channel can be activated from DN keys or two-way HOT line keys where the DN for the HOT line is a mixed appearance with a DN of a speech monitor system channel. The user makes a call from this specific DN or HOT line key. When the call is established the user then puts the call on hold by using automatic hold or manual hold. The corresponding channel on the speech monitor system will automatically be activated. The call can then be heard on the speech monitor unit when the channel is selected. At any time the user can enter the call on the speech monitor unit by using the built-in microphone (if provided) and this two-way conversation will be heard on the loudspeaker in addition to any other channels active on the loudspeaker.

To talk privately to one of the calls being monitored on the speech monitor unit, the user takes the call on the handset of the phone. This conversation will not be heard on the loudspeaker, but any other user with the same DN appearance will be able to enter the call by going off-hook and establishing a multiple appearance conference.

If the user presses the Release key while active on a call that appears on a speech monitor system channel, the call is disconnected from all DN appearances, including the speech monitor system channel.

It is not possible to prevent the speech monitor unit from becoming active. If a user no longer wishes to listen to the speech monitor, the unit needs to be switched off manually.

Introduced in X11 Release:	All
Networking:	No

On-Hook Dialing

The On-Hook Dialing feature enables a Meridian 1 proprietary telephone user to make a call without lifting the handset. Signaling tones and the voice of the called party are heard over the loudspeaker. For two-way communication, the user must lift the handset or activate the Handsfree unit if equipped.

Operating parameters

The On-Hook Dialing feature does not apply to analog (500/2500 type) telephones.

Feature interactions

LOGIVOX Telephone

Because of the firmware on the LOGIVOX set, the DN key 0 is automatically selected when the first digit is dialed, and no other DN has been selected.

Feature packaging

On-Hook Dialing is included in base X11 system software.

Feature implementation

No change to existing configuration is required for the On-Hook Dialing feature.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	5
Networking:	No

Optional Outpulsing Delay

The Optional Outpulsing Delay (OOD) feature increases to three seconds the Start of Dialing Delay used for automated dialing on loop start Central Office (CO) trunks. This feature is required for Meridian 1 connection in some countries.

Operating parameters

There are no feature requirements.

Feature interactions

Features that automatically dial digits onto a loop start CO trunk are provided with an additional delay. These features include the following:

- Stored Number Redial
- Autodial
- Speed Call
- Call Forward All Calls
- Basic Alternate Route Selection/Network Alternate Route Selection (BARS/NARS)
- System Speed Call, System
- Network Speed Call, and
- Flexible Hot Line.

Feature packaging

Optional Outpulsing Delay (OOD) package 79 has no feature package dependencies.

Feature implementation

No change to existing configuration is required for the Optional Outpulsing Delay feature.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	20.0x
Networking:	No

Out-of-Service Unit (OOSU)

The ability to mark a unit as "Out of Service" is a feature that is part of the Global Line Cards program. This capability is accomplished through Service Change. A unit marked Out of Service cannot be configured as any other type of unit without first removing it from the Out-of-Service state. A unit marked Out of Service stays Out of Service through Initialization or SYSLOAD operation. This feature reduces the number of cards that must be replaced in situations where only one, or a few circuits, fails to work in the field. In addition, the capability enables support personnel to change high density cards at convenient low-traffic periods.

Operating parameters

There are no feature requirements.

Feature interactions

There are no interactions with other features.

Feature packaging

No new software packages have been introduced for this feature.

Feature implementation

LD 10 – The new response OOSSLT is added to the TYPE prompt for designating single-line terminal units as Out of Service.

It provides the capability to designate an analog (500/2500 type) telephone as being Out of Service, regardless of card density. This Out-of-Service status survives a SYSLOAD. To reconfigure a unit as another type of unit it is necessary to first remove the unit from its Out-of-Service status, and then reconfigure it as NEW.

Prompt	Response	Description
REQ	NEW OUT	New, or remove.
TYPE	OOSSLT	Out-of-service single-line terminal unit.
TN	l s c u c u	Terminal Number. For Option 11C. If the REQ is NEW, a check is made to verify that the card already exists, and the unit specified is not already configured. If the REQ is OUT, a check is made to verify that the unit is marked Out of Service. If the unit specified to be removed is the last configured unit on the card, the card blocks associated with the logical card are removed.

LD 11 – A new prompt OOSMLT is added for designating multi-line terminal units Out of Service, and provides the capability to make any unit Out of Service, regardless of the card type or density.

Prompt	Response	Description
REQ	NEW OUT	New, or remove.
TYPE	OOSMLT	Out of Service multi-line terminal unit.
TN	l s c u c u	Terminal Number. For Option 11C. If the REQ is NEW, a check is made to verify that the card already exists, and the unit specified is not already configured. If the REQ is OUT, a check is made to verify that the unit is Out of Service. If the unit specified to be removed is the last configured unit on the card, the card blocks associated with the logical card are removed.

Feature operation

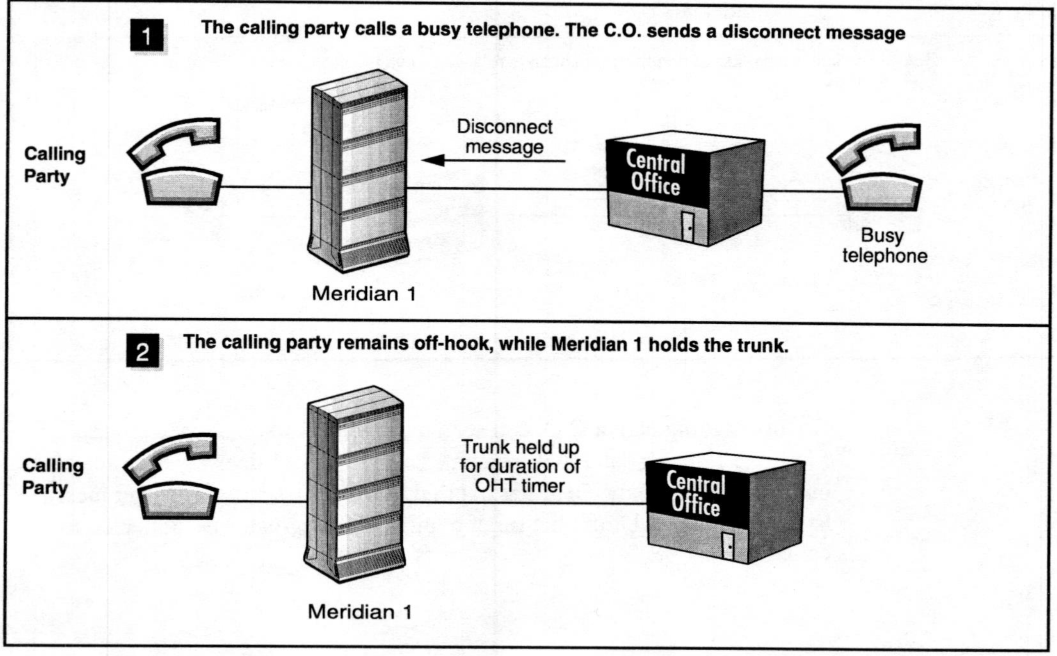
No specific operating procedures are required to use this feature.

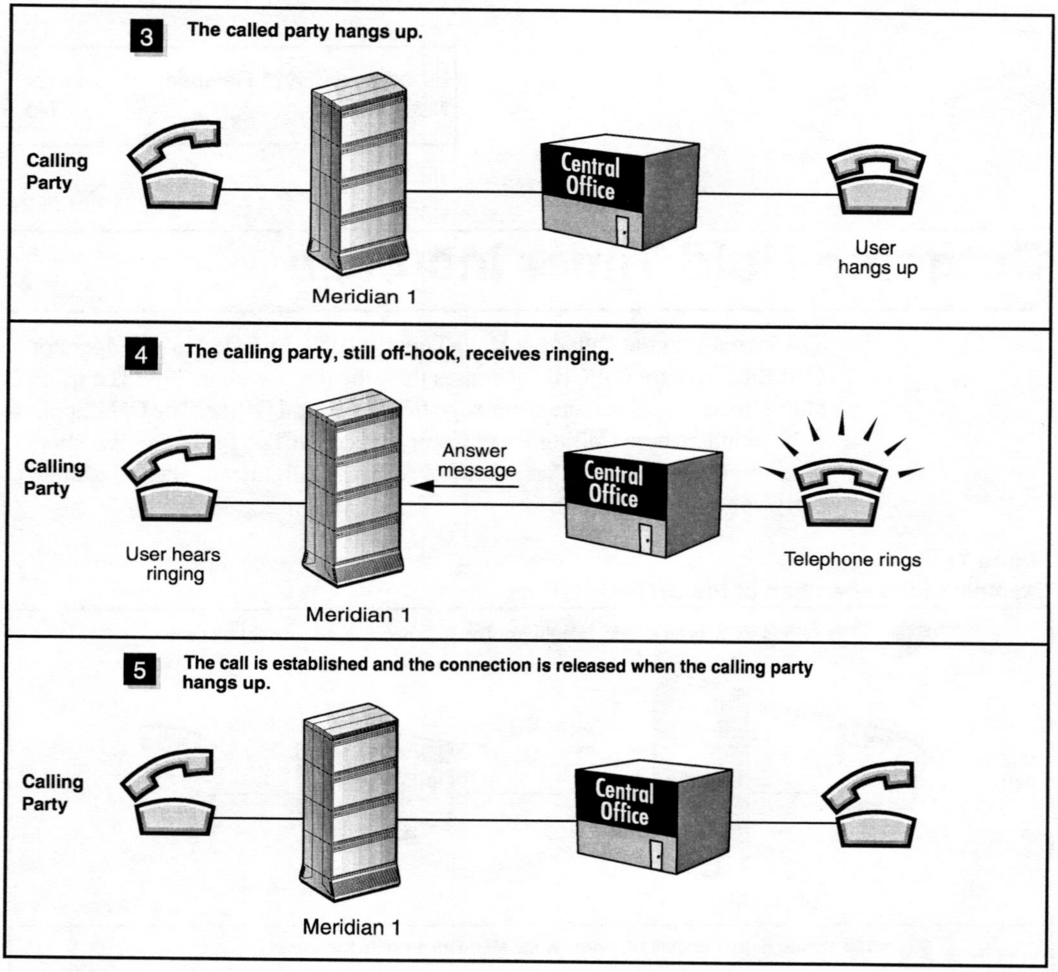
Introduced in X11 Release:	24.2x
Networking:	No

Outgoing Hold Timer Increase

The increase to the Outgoing Hold Timer (OHT), included in the Operator Call Back feature (OPCB), increases the time the Meridian 1 holds a trunk after it receives a disconnect message from a Central Office. The OHT applies to situations where Calling Party Control is active.The following is a short description of a Calling Party Control (CGPC) call, (a call where the calling party controls the disconnect).

Figure 71
Example of the operation of the OHT





On an outgoing call, a C.O. can send a disconnect message back to the Meridian 1 during call establishment. The Meridian 1 does not disconnect the outgoing call until the OHT has expired. If the C.O. sends an answer message to the Meridian 1 before the timer expires, the originator is connected to the called party.

The OHT determines the length of time the Meridian 1 holds a trunk after receiving a disconnect message.The maximum is now 126 seconds. The timer is programmed in increments of 2 seconds. Before this update, the OHT expired after a maximum of 62 seconds. The default value of 30 seconds has not changed.

Operating Parameters

This feature enhances the existing OHT capability provided by Package 126.

The OPCB OHT is available on analog and DTI2 trunk interfaces. It is not supported on DTI1.5 trunk routes.

Feature Interactions

Outgoing Hold Toll Timer

When the C.O. sends a disconnect message on an outgoing toll call, the Outgoing Hold Toll Timer (OHTT) disconnects after a maximum of 90 seconds. The OHTT can be programmed in increments of two seconds. There has been no change to this feature.

Feature Packaging

No new packages have been introduced for this feature. Operator Call Back (OPCB) package 126 is required.

Feature Implementation

LD 16 – Configure the OHT on the trunk route.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	RDB	Route Data Block.
ROUT	0-511	Route Number.
CNTL	YES	Changes to control or timers.
...		
OPCB	YES	Enable the Operator Call Back feature.

...		
OHT	0-(30)-126	Outgoing Hold Timer in seconds (programmed in increments of two seconds).
. . .		

Feature Operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	18
Networking:	No

Overlay Cache Memory

With X11 Release 18 and later, Overlay Cache Memory uses Protected Data Storage (PDS) as a cache area for storing overlays loaded from disk. The cache memory overlays are accessed much faster than those on disk, reducing the load time to approximately one second.

A maximum of 32 overlays can reside in Overlay Cache Memory at one time. The CACH prompt in LD 17 defines the number of cache memory buffers allocated in protected memory. Each overlay resides in a buffer. A zero entry deactivates this feature and requires all overlays to be loaded from disk.

Each buffer requires 19K of Protected Data Storage (PDS). If there is insufficient memory to store the number of buffers requested, a warning message follows the LD 17 prompt sequence. The message indicates that more memory is required to store all the caches requested.

If a small number of cache memory buffers are allocated, frequently used overlays may be removed from protected memory by seldom used overlays. The PRTY prompt in LD 17 sets an overlay priority flag. A priority flag prevents the removal of an overlay from cache memory. The number of priority flags set cannot exceed the number of cache memory buffers specified.

When an LD xx command is entered, the cache memory is checked for the requested overlay. If the requested overlay is in cache memory, its data portion is rapidly copied to the regular overlay area.

A requested overlay that is not in cache memory is loaded from the disk into the normal overlay area and simultaneously stored into a cache memory buffer, if one is available. If one is not available, the new overlay overwrites another in the cache memory.

If an overlay is loaded from disk and no unused buffer area exists, the overlay used longest ago without its priority flag set is removed and replaced by the new overlay.

Operating parameters

If the feature is deactivated with a zero (0) entry at the CACH prompt in LD 17, no cache memory exists and all overlays are loaded from disk.

Cache memory is not affected by a system initialization. After a system initialization, it is not necessary to reload overlays from the disk.

Each buffer requires 19K of PDS. The number of cache memory buffers allocated to the system is limited by the availability of spare memory. If enough memory exists, a maximum of 32 cache memory buffers is allowed. Each buffer stores one overlay.

The number of overlay priorities (PRTY) that can be set is dependent upon Release.

— Release 18 - The number of overlay priorities cannot exceed the number of cache memory buffers allocated.

— Release 19 and 20 - The number of cache memory buffers (CACH) allocated must exceed the number of overlay priorities (PRTY) by three. (CACH = PRTY + 3)

— Release 21 and later - The number of cache memory buffers (CACH) allocated must exceed the number of overlay priorities (PRTY) by four. (CACH = PRTY + 4)

To load an overlay from disk, use the command LD xx D. This is necessary for the system to determine which overlay to read. The LD xx D command loads the overlay from disk and overwrites the same overlay existing in cache memory.

Using the LD xx D command to force load an overlay from disk does not simultaneously support the peripheral download SUSP command.

When overlays are stored in cache memory, the ENLT and DIST commands are not supported.

The system automatically stores and retrieves overlays from cache memory. If the cache area is full when a new overlay is requested, the overlay gone unused the longest without a priority flag set is removed and replaced by the new overlay. Daily routines and background-loaded overlays are not stored in cache memory.

The Overlay Cache Memory feature does not apply to Option 51C, 61C, 81, and 81C systems.

Conversion and upgrades

Due to memory requirements, installing a new issue of software or the same issue with additional features may reduce the number of cache buffers that can be allocated. A warning message indicates this reduction has occurred.

If this reduction causes the number of overlay priorities to exceed the maximum number of cache buffers, the overlay priorities are reduced to equal the number of cache buffers. The priorities are automatically reduced by beginning with the highest overlay number and working downward.

Feature packaging

Overlay Cache Memory is included in the base X11 system software.

Feature implementation

LD 17 – Change system configuration record.

Prompt	Response	Description
REQ	CHG	Change data.
TYPE	CFN OVLY	Configuration Record. Release 19 gate opener.
OVLY	YES	Change overlay area.
- CACH	(0) 2-32	Number of overlay buffers held in cache memory. Entering 0 disables the feature.
- PRTY	xx xx xx xx...	Set priority for the stored overlays. Priority can be set only for the number of overlays specified in CACH. xx = the overlay number. An X preceding the number deletes the priority flag for that overlay.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	16.90G
Networking:	No

Overlay 45 Limited Repeats

Overlay or LD 45, the Background Continuity Diagnostics, is automatically loaded whenever a power fault is detected, and run in the background. This feature allows a limit to be placed on the number of times that background continuity tests are run by this overlay. This limit is system configured in LD 17, and may have a value from 0-31. Once the defined value has been reached, the regular background programs are restored. The alarm is not cleared. Since the alarm has not been cleared, LD 45 is not reloaded before the end of the current midnight routine cycle. At the end of the midnight cycle, the alarm is cleared by the overlay supervisor.

If there are no midnight routines, LD 45 starts a timer which is decreased at regular intervals by the work scheduler. The alarm is not cleared at this point. Therefore, LD 45 is not reloaded for an alarm condition. When the timer expires, the work scheduler clears the alarm. If another alarm condition arises, LD 45 is automatically loaded and runs as described above.

Operating parameters

The system overlay loader checks for power alarms and sets the relevant task request bit, if found. This overlay loader is modified to ignore power alarms once the limit defined for the overlay repeats has been reached, until the end of the current midnight routine cycle, if there is one.

It is advised that a printer be used to obtain hard copy information on the continuity tests run by LD 45, rather than relying on the history file, if one is available.

Feature interactions

There are no interactions with other features.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 17 – Configure Overlay 45 Limited Repeats parameters.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	CFN OVLY	Configuration Record. Release 19 gate opener.
...		
- CY45	(0)-31	Cycles of LD 45. Cycles of LD 45 can be run whenever a fault is detected. If any number from 1 to 31 is entered, that is the number of times LD 45 will run under fault conditions. If 0 is entered, the system will perform as before without limiting the number of LD 45 runs.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	15.58F
Networking:	No

Override

The Override feature provided in base X11 system software allows a user to enter into an established connection. A warning tone notifies the talking parties that a third party is about to enter the conversation. The warning tone is an initial one-second burst, followed by a 256 millisecond burst repeated every 16 seconds. The Override feature can be used after a user has dialed a busy Directory Number (DN).

Forced Camp-On and Priority Override

With X11 Release 20 and later software, the Forced Camp-On and Priority Override features provide enhancements to the basic Camp-On feature. Forced Camp-On is similar to the regular Station-to-Station Camp-On, except that it can be done without an internal or external call on hold. Forced Camp-On is activated automatically (if Automatic Forced Camp-On is defined); or it can be activated manually using the Enhanced Override (EOVR) key on Meridian 1 telephones or the Enhanced Override Flexible Feature Code on analog (500/2500 type) telephones.

Priority Override allows an established call to be broken into and another call to be presented to the desired party. Before break-in occurs, a warning tone is given to all parties involved in the established call. The telephone performing the override must have a priority level equal to or higher than the telephone being overridden. Priority Override is activated by dialing the Override Flexible Feature Code on analog (500/2500 type) telephones, or by pressing the Override key (OVR) on Meridian 1 telephones.

For detailed information on the Forced Camp-On and Priority Override features, refer to the Camp-On, Forced, and Override, Enhanced feature descriptions contained in this document.

Operating parameters

On Meridian 1 proprietary telephones, a separate Override key must be assigned. An associated lamp is not required.

On analog (500/2500 type) telephones, a Flexible Feature Code (FFC) is required to override a call.

Override cannot be used to enter an established connection if any party (telephone or trunk) has Warning Tone Denied Class of Service. In this case, overflow tone is heard.

The system must have a conference loop.

Feature interactions

Attendant Break-In

When one Meridian 1 telephone has overridden an existing call to establish a Conference call, Break-In is temporarily denied. The attendant is notified via the Override tone.

Telephones with a toll operator break-in call cannot be overridden. Overflow tone is returned to telephones attempting Priority Override.

Automatic Redial

An Automatic Redial (ARDL) call cannot be overridden. This is done to avoid creating a conference when a tone detector is involved.

Call Forward/Hunt Override Via Flexible Feature Code

It is possible to use Priority Override after using the Call Forward/Hunt Override FFC and encountering a busy set.

Call Party Name Display

When Overriding an established call, the displays of the other telephones show the DN and name of the overriding party.

Camp-On

Station-to-Station Camp-On and Attendant Camp-On are not affected by Forced Camp-On or Priority Override. The following new Classes of Service affect only Forced Camp-On:

— Camp-On From Another Telephone Allowed (CPFA)

— Camp-On From Another Telephone Denied (CPFD)

— Camp-On To Another Telephone Allowed (CPTA)

— Camp-On To Another Telephone Denied (CPTD)

The Station Camp-On (SCMP) package 121 is required to return busy tone instead of ringback tone to the party camping on.

Camp-On, Forced

When Priority Override is activated, it replaces normal override. Once Priority Override has been performed on a set, its Digit Display shows the DN of the overriding set.

Charge Account and Calling Party Number

When Charge Account is used during active Override, some digits may be lost. When entered with Override in conference, a Charge Account number is accepted and no digits are lost.

China – Attendant Monitor

A set may operate override to join into a desired call. If the desired call is being Attendant Monitored at the time, one of the following occurs:

— If the desired call is a conference call, the override attempt is blocked as per existing operation.

— If the call is a simple one with the Attendant Monitoring with no tone, the override attempt is successful and Attendant Monitor is deactivated.

— If the call is a simple one with the Attendant Monitoring with tone, the override attempt is blocked.

Conference

Override cannot be used to enter a Conference call.

Do Not Disturb

Telephones with Do Not Disturb enabled cannot be overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.

Group Hunt

Override will not be supported.

Hot Line

A Hot Line call can be entered using the Override feature.

Make Set Busy

Telephones with MSB active cannot be overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.Voice Call is blocked by MSB.

Multi-Party Operations

With Multi-Party Operations (MPO), when a consultation call is made on a set equipped with Priority Override, a control digit has to be dialed from the set to perform a recall and return the call on hold.

Network Intercom

An internal Hot Type I call never returns busy, unless the call became a non-Hot Line call due to the Hot Line key being busy. In this case, the call behaves like a normally dialed call, and override can be used upon receipt of a busy signal.

Night Restriction Classes of Service

If Priority Override and Night Restriction for Priority Override Class of Service (NROA) are assigned, Priority Override will be operational for the set only when Night Service is in effect.

On Hold on Loudspeaker

It will not be possible to Override into a call on loudspeaker as it is effectively on hold at the set.

Override, Enhanced

If Priority Override is equipped, it replaces Override when using the OVR key or OVRD FFC. However, Override can be simulated by using the default PLEV, 2, for all trunk routes and sets.

Periodic Camp-on Tone

The Periodic Camp-On Tone has precedence over Override intrusion tone.

Phantom Terminal Numbers
Call Forward

Call Forward cannot be overridden on phantom terminal numbers. The overflow tone occurs if an Override is attempted.

Ring Again

Ring Again is the only other feature currently available once a busy telephone has been encountered. Ring Again is not allowed on an analog (500/2500 type) telephone making a Multi-Party Operations consultation call.

Uninterrupted Line Connections

Override cannot be applied to stations with a Warning Tone Denied Class of Service.

Feature packaging

Override is included in base X11 system software.

For analog (500/2500 type) telephones, Flexible Feature Code (FFC) package 139 must be equipped.

Forced Camp-On/Priority Override (POVR) is package 186.

Feature implementation

LD 10 – Allow Override for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	(OVDD) OVDA (XFD) XFA (WTA) WTD	Override (denied) allowed for this telephone. Transfer (denied) allowed. Warning Tone (allowed) denied (WTA is required to be overridden).

LD 11 – Add or change Override for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = Meridian 1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	(WTA) WTD	Warning Tone (allowed) denied (WTA is required to be overridden).
KEY	xx OVR	Override key (must be key 34 for the M3000).

LD 14 – Define Warning Tone Allowed for trunks to permit Override.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaa	Trunk type, where: aaa = ADM, AID, ATVN, AWR, CAA, CAM, COT, CSA, DIC, DID, FEX, ISA, MDM, MUS, PAG, RAN, RCD, RLM, RLR, TIE, or WAT.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	(WTA) WTD	Warning Tone (allowed) denied (WTA is required to be overridden).

LD 57 – Configure Flexible Feature Code (FFC) for Override on analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FFC	Flexible Feature Codes.
CUST	0-99 0-31	Customer number. For Option 11C.
CODE	OVRD	Change Override access code.
OVRD	xxxx	Override access code.

Feature operation

To override a call in progress from a Meridian 1 proprietary telephone:

1 Dial the number. You hear a busy tone.

2 Press **Override**. Everyone hears a one-second tone burst.

3 You are connected to the call.

To cancel Override from a Meridian 1 proprietary telephone:

1 Press **Rls** or hang up.

2 You are disconnected. The original call remains active.

To override a call in progress from a analog (500/2500 type) telephone:

1 Dial the number. You hear busy tone.

2 Flash the switchhook or press **LINK**.

3 Dial the Flexible Feature Code (FFC) for Override. Everyone hears a one-second tone burst.

4 You are connected to the call.

To cancel Override from a analog (500/2500 type) telephone:

1 Press **Rls** or hang up.

2 You are disconnected. The original call remains active.

Introduced in X11 Release:	20
Networking:	No

Override, Enhanced

The use of the Forced Camp-On and Priority Override features together results in Enhanced Override (EOVR).

Forced Camp-On

Forced Camp-On allows a call to be camped on and a warning to be given before the Priority Override operation. It differs from normal Camp-On in that both internal and external calls can be camped on, rather than just external calls as with the Camp-On feature. The Forced Camp-On may be automatic or manual. The manual operation requires the use of the Enhanced Override (EOVR) feature.

Forced Camp-On can be used as a feature by itself or in conjunction with Priority Override. The combination of the two features is referred to as Enhanced Override (EOVR).

For manual Forced Camp-On an analog (500/2500 type) telephone, the user has to dial the EOVR Flexible Feature Code (FFC), while a Meridian 1 proprietary telephone user has to use the EOVR key.

A second operation of the EOVR key or FFC executes Enhanced Override.

Forced Camp-On is similar to Station-to-Station Camp-On, except that Forced Camp-On can be done with either no call on hold or an external or internal call on hold. It can be done automatically or manually, which is determined by the response to the Automatic Forced Camp-On (AFCO) prompt in LD 15.

For manual operation, once a busy telephone has been reached, the first depression of the EOVR key or the first dialing of the EOVR FFC attempts Forced Camp-On. If successful, Forced Camp-On introduces Camp-On tone into the connection. If unsuccessful, overflow (fast busy) tone is returned to the party attempting the Forced Camp-On.

For Forced Camp-On to be allowed, all other methods of call termination must have been tried, and the last one must be Camp-On. If Station-to-Station Camp-On or Automatic Forced Camp-On has occurred, or Forced Camp-On has been excluded by the new telephone options, then the first depression of the EOVR key or first dialing of the EOVR FFC introduces Enhanced Override. If, however, Forced Camp-On is denied by existing Camp-On restrictions, Enhanced Override is also denied.

Priority Override

The Priority Override (POVR) feature allows users to break in to an established connection. To do this, analog (500/2500 type) telephone users use the OVRD Flexible Feature Code (FFC), and Meridian 1 proprietary telephone users use the Override (OVR) key before Camp-On.

The Priority Override Level (PLEV) restrictions apply to both Enhanced and Priority Override.

For Priority Override, the overriding telephone must have a Priority Override Level (PLEV) greater than or equal to the PLEV of the telephone or trunk to be overridden.

For an analog (500/2500 type) telephone, a recall followed by dialing the Priority Override FFC, (Override FFC with Priority Override package 186 equipped), breaks into the connection and establishes a conference between all three parties and sends an override tone. For a Meridian 1 proprietary telephone, the OVR key is used in place of the FFC.

In order for Priority Override to be allowed, all telephones and trunks involved must have Warning Tone Allowed (WTA) Class of Service. Each telephone and trunk route (TIE, DID, and COT) is assigned a PLEV where:

PLEV 0	Indicates that this telephone or route cannot be overridden; if assigned to a telephone, the telephone cannot override.
PLEV 1	Indicates that this telephone or route can be overridden; if assigned to a telephone, the telephone cannot override.
PLEV 2	Indicates that this telephone or route can be overridden by telephones assigned level 2 through level 7; if assigned to a telephone, the telephone can override level 1 and level 2.
PLEV 3-6	Similar to level 2, indicates that this telephone or route can be overridden by telephones assigned an equal or higher level; if assigned to a telephone, the telephone can override lower and equal levels, except level 0.
PLEV 7	Indicates that this telephone or route can be overridden by another level 7 telephone only; if assigned to a telephone, the telephone can override level 1 through level 7.

Several combinations of the Forced Camp-On and Priority Override are highlighted in the following list:

— Responding to the Automatic Forced Camp-On (AFCO) prompt in LD 15 with “NO,” configuring Meridian 1 proprietary telephones with only Override (OVR) keys, and defining the Override (OVRD) Flexible Feature Code (FFC) disallows the use of Forced Camp-On.

— Responding to the Automatic Forced Camp-On (AFCO) prompt in LD 15 with “NO” and setting the Priority Level (PLEV) to 0 and the Camp-On Classes of Service to Camp-On From Another Telephone Denied (CPFD) and Camp-On To Another Telephone Denied (CPTD) gives manual Camp-On only.

— Configuring the EOVR FFC for analog (500/2500 type) telephone users and equipping Meridian 1 proprietary telephones with EOVR keys gives the users the ability to use only Priority Override (via OVR key or OVRD FFC) or Forced Camp-On followed by Priority Override (pressing the EOVR key twice or using EOVR FFC).

— Responding to the Automatic Forced Camp-On (AFCO) prompt in LD 15 with "YES," configuring Meridian 1 proprietary telephones with only Override (OVR) keys, and defining the Override (OVRD) Flexible Feature Code (FFC) automatically applies Forced Camp-On in situations where it is allowed, and allows the use of the OVR key and FFC to implement Priority Override.

— Using an EOVR key or FFC with a response of "YES" to the AFCO prompt in LD 15 simulates the Override (OVR) key or FFC unless Forced Camp-On was denied initially, in which case the Forced Camp-On would be re-attempted.

The following table summarizes the various combinations:

	AFCO = NO	AFCO = YES
OVR FFC or key	Attempts Priority Override.	Attempts Priority Override whether Forced Camp-On occurred or not.
EOVR FFC or key	First use attempts Forced Camp-On, unless station is camped on, then Priority Override is attempted. Second use attempts Priority Override.	If automatic Forced Camp-On was denied, re-attempts Forced Camp-On; otherwise Priority Override is attempted.

Operating parameters

The Flexible Feature Codes (FFC) package 139 must be equipped for Forced Camp-On and Priority Override to be available from analog (500/2500 type) telephones.

For analog (500/2500 type) telephone activation, the Multi-Party Operations (MPO) package 141 must be equipped, with the "YES" as the response to the RALL prompt in LD 15 to ensure that register recalls are required before dialing control digits. The OVRD and EOVF FFCs defined must not start with the same digit as one of the control digits. The control digits are defined in LD 15 and are printed as part of the Customer Data Block (LD 21).

If Priority Override is equipped, it replaces Override when the user uses the OVR key or OVRD FFC. However, Override can be simulated by using the default value, 2, for all trunk routes and telephones.

Telephones or trunks involved in any of the following cannot be camped on or overridden:

- Non-established call
- Conference call
- Attendant call
- Attendant call via Centralized Attendant Service (CAS), Primary Rate Interface (PRI), or Integrated Services Digital Network (ISDN) trunk
- Make Set Busy
- Do Not Disturb
- Automatic Call Distribution (ACD) call
- Operator Call Back
- Hold
- Data call
- Release Link call, or
- Parked call.

Call Forward and Hunting take precedence over Call Waiting. If Call Waiting is allowed, Camp-On is not attempted. If Call Waiting is not allowed, Station-to-Station Camp-On is automatically attempted. If this succeeds, Priority Override can still follow. If Camp-On fails because there is no external call, Forced Camp-On and Priority Override may still work. However, if Camp-On fails because of other limitations, Forced Camp-On and Priority Override will also not work.

Even though Camp-On will still function when Warning Tone Denied (WTD) Class of Service is defined, Forced Camp-On and Priority Override require Warning Tone Allowed (WTA) Class of Service.

Priority Override is not allowed on analog (500/2500 type) telephones unless the Override Allowed (OVDA) Class of Service is defined. This Class of Service is also used for Override. This Class of Service does not affect Camp-On.

Camp-On requires an external call on hold. Forced Camp-On can be done without a call on hold, or with both internal or external calls on hold.

Trunks cannot perform Priority Override. They also cannot be overridden unless they are the unwanted party of a connection. It is for this exception that trunks are given a Priority Level.

New Camp-On Classes of Service (Camp-On From Another Telephone Allowed [CPFA], Camp-On From Another Telephone Denied [CPFD], Camp-On To Another Telephone Allowed [CPTA], and Camp-On To Another Telephone Denied [CPTD]) apply to Forced Camp-On and Automatic Forced Camp-On (AFCO) only. They do not apply to Station or Attendant Camp-On.

If a telephone is denied Forced Camp-On by Class of Service, Priority Override may still be attempted.

Feature interactions

Attendant Break-In

Telephones with a toll operator break-in call cannot be camped on to or overridden. Overflow tone is returned to telephones attempting either Forced Camp-on or Priority Override.

Attendant calls

Telephones involved in attendant calls cannot be force camped on or priority overridden. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-On or Priority Override.

Automatic Call Distribution

Telephones involved in Automatic Call Distribution (ACD) calls cannot be force camped on or priority overridden. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-On or Priority Override.

Call Hold, Deluxe
Call Hold, Permanent

Neither held calls nor telephones with calls on hold may be camped on or overridden. Overflow (fast busy) tone is returned to telephones attempting either a Forced Camp-On or Priority Override.

Camp-On

Station-to-Station Camp-On and Attendant Camp-On are not affected by Forced Camp-On or Priority Override. The following new Classes of Service affect only Forced Camp-On:

— Camp-On From Another Telephone Allowed (CPFA)

— Camp-On From Another Telephone Denied (CPFD)

— Camp-On To Another Telephone Allowed (CPTA)

— Camp-On To Another Telephone Denied (CPTD])

The Station Camp-On (SCMP) package 121 is required to return busy tone instead of ringback tone to the party camping on.

Conference calls

Telephones involved in conference calls cannot be force camped on or priority overridden. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-On or Priority Override.

China – Attendant Monitor

A set may operate enhanced override on a desired call. If the desired call is being Attendant Monitored at the time, existing operation occurs for the first time the Enhanced Override key is pressed. The second time the key is pressed, the interaction with Attendant Monitor is the same as with regular override.

Data calls

Data calls have Warning Tone Denied (WTD) Class of Service, and cannot be force camped on or priority overridden. Overflow (fast busy) tone is returned to telephones attempting Forced Camp-On or Priority Override.

Digit Display

The Digit Display of the telephones being overridden changes to the Directory Number (DN) of the telephone overriding once Priority Override is accomplished.

Do Not Disturb

Telephones with Do Not Disturb (DND) enabled cannot be force camped on or priority overridden. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-On or Priority Override.

Hold

Neither held calls nor telephones with calls on hold may be camped on or overridden. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-On or Priority Override.

Make Set Busy

Telephones with Make Set Busy active cannot be Forced Camp-On or Priority Override. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-On or Priority Override.

Multi-Party Operations

With Priority Override (POVR) equipped, there is a slight change in Multi-Party Operations functionality. When a consultation call is made without POVR equipped, and the telephone being called is busy, a recall returns to the party on hold without dialing a control digit. However, if POVR is equipped, a control digit must be dialed. Any control digit releases the busy call and returns to the call on hold.

Operator Call Back

Telephones involved in an Operator Call Back call or Toll Operator Break-In cannot be force camped on or priority overridden. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-On or Priority Override.

Override

If Priority Override is equipped, it replaces Override when using the OVR key or OVRD FFC. However, Override can be simulated by using the default PLEV, 2, for all trunk routes and telephones.

Ring Again

Ring Again (RGA) is the only other feature currently available once a busy telephone has been encountered. RGA is not allowed on an analog (500/2500 type) telephone making a Multi-Party Operations consultation call.

Feature packaging

To provide the Enhanced Override capabilities, the following packages are required:

— Station Camp-On (SCMP) package 121

— Flexible Feature Codes (FFC) package 139

— Multi-Party Operations (MPO) package 141, and

— Priority Override/Forced Camp-On (POVR) package 186.

Feature implementation

LD 15 – Configure the customer for Automatic Forced Camp-On and Station Camp-On tone.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	CDB MPO	Customer Data Block. Release 21 gate opener.
...		
- AFCO	(NO) YES	Automatic Forced Camp-On. Enter YES if Forced Camp-On is to be applied automatically. Enter NO if Forced Camp-On is to be applied manually.
TYPE	FTR	Release 21 gate opener.
...		

- STCB	(NO) YES	Station Camp-On Busy tone. Enter NO if Busy Tone is not to be given to the transferring (controlling) party when the desired station is busy. Enter YES if Busy Tone is to be given to the transferring (controlling) party when the desired station is busy.

LD 57 – Configure Override and Enhanced Override FFCs.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add, or change.
TYPE	FFC	Flexible Feature Codes.
...		
CODE	x...x	Code to be programmed. Where x...x may be one of the following: OVRDOverride (OVRD is used for Priority Override when the Priority Override [POVR] package 186 is equipped.) EOVREnhanced Override (Is programmable only when the Priority Override [POVR] package 186 is equipped.)
X...X	y...y	The user is prompted with X...X, where X...X is the FFC code entered in response to the CODE prompt. y...y is a one-to-seven character input that the user must dial to use the FFC. Valid inputs are digits 0 through 9, asterisk (*), and octothorpe (#).

LD 10 – Configure analog (500/2500 type) telephones for Forced Camp-On, Priority, and Enhanced Override.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	500	Telephone type.
...		
CLS		Class of Service.
	(CPFA) CPFD	Forced Camp-On from another telephone to this telephone (allowed) denied.
	(CPTA) CPTD	Forced Camp-On to another telephone from this telephone (allowed) denied.
	OVDA	Override allowed.
	WTA	Warning Tone allowed.
...		
PLEV	0-(2)-7	Priority Override Level 0Indicates that this telephone cannot be overridden or override. 1Indicates that this telephone can be overridden but cannot override. 2Indicates that this telephone can be overridden by telephones assigned level 2 through level 7 and that the telephone can override level 1 and level 2. 3-6Similar to level 2, indicates that this telephone can be overridden by telephones assigned an equal or higher level and that it can override lower and equal levels, except level 0. 7Indicates that this telephone can be overridden by another level 7 telephone only and that it can override level 1 through level 7.

LD 11 – Configure Meridian 1 proprietary telephones for Forced Camp-On, Priority, and Enhanced Override.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	aaaa	Telephone type, where: aaaa = Meridian 1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
...		
CLS		Class of Service.
	(CPFA) CPFD	Forced Camp-On from another telephone to this telephone (allowed) denied.
	(CPTA) CPTD	Forced Camp-On to another telephone from this telephone (allowed) denied.
	WTA	Warning Tone allowed.
...		
PLEV	0-(2)-7	Priority Override Level. 0 Indicates that this telephone cannot be overridden or override. 1Indicates that this telephone can be overridden but cannot override. 2Indicates that this telephone can be overridden by telephones assigned level 2 through level 7 and that the telephone can override level 1 and level 2. 3-6Similar to level 2, indicates that this telephone can be overridden by telephones assigned an equal or higher level and that it can override lower and equal levels, except level 0. 7Indicates that this telephone can be overridden by another level 7 telephone only and that it can override level 1 through level 7.

...		
KEY		Define keys.
	xx OVR	Override (If Priority Override [POVR] package [186] is equipped, the OVR key is used for Priority Override.)
	xx EOVR	Enhanced Override (Allowed to be programmed only if Priority Override [POVR] package [186] is equipped.)

LD 16 – Configure Route for Forced Camp-On, Priority, and Enhanced Override.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block
...		
PLEV	0-(2)-7	Priority Override Level. 0Cannot be overridden. 1-7Can be overridden by a telephone with a Priority Level that is equal to or greater than the level assigned to this route. ***Note:*** Trunks cannot override, but the levels of all parties in a connection are examined to determine if the connection may be overridden.

LD 14 – Configure trunks for Forced Camp-On, Priority, and Enhanced Override.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
...		
CLS		Class of Service.
	WTA	Warning Tone Allowed.

Feature operation

Forced Camp-On and Priority Override can be used when making either a simple call or consultation call (i.e., having a call on hold while calling another party) call. The following feature operation descriptions use telephone A (an analog (500/2500 type) telephone) or telephone E (a Meridian 1 proprietary telephone) to call telephone B, which is connected to party C. Party D is used as the party on hold when either A or E is making a consultation call.

The telephones are configured as follows:

1 Telephone A is an analog (500/2500 type) telephone with Warning Tone Allowed (WTA) and Override Allowed (OVDA) Classes of Service.

2 Telephone B has Warning Tone Allowed (WTA) Class of Service.

3 Party C has Warning Tone Allowed (WTA) Class of Service and can be any telephone type or a Direct Inward Dial (DID), TIE, or Central Office (Public Exchange) (COT) trunk.

4 Party D can be any telephone or trunk.

5 Telephone E is a Meridian 1 proprietary telephone with Warning Tone Allowed (WTA) Class of Service and both an Override (OVR) and Enhanced Override (EOVR) key equipped.

For examples 1 to 4, assume the following:

1 Telephones A and E have a Priority Override Level (PLEV) of greater than "1".

2 Telephones A and E both have Camp-On From Another Telephone Allowed (CPFA) Class of Service.

3 Telephone B and party C both have PLEVs greater than "0", but less than or equal to those of telephones A and E.

4 Both telephone B and party C are involved in a simple call, not a conference call.

5 Telephone B has Camp-On To Another Telephone Allowed (CPTA) Class of Service.

6 Call Forward, Hunting, and Call Waiting are not in use.

Examples 1 to 4 are done with various combinations of Forced Camp-On and Priority Override. Forced Camp-On may be denied by responding "NO" to the Automatic Forced Camp-On (AFCO) prompt in LD 15, by configuring telephone E with only an Override (OVR) key and defining only the Override (OVRD) FFC in LD 57, or by setting the Classes of Service for both telephone A and E to Camp-On To Another Telephone Denied (CPTD) and Camp-On From Another Telephone Denied (CPFD). Both of these methods of disabling the Forced Camp-On feature do not affect the Priority Override feature. However, any conditions that would prevent Forced Camp-On from occurring also prevent Priority Override.

In the following feature operation descriptions, the term "recall" refers to a register recall, which may be performed in a number of different ways. Some typical examples are:

— Flashing the switchhook. This is the equivalent of hanging up the handset and picking it back up. This on hook, off hook action is performed in a time less than what the system would consider to be a valid disconnect.

— Pressing the flash or LINK button if equipped.

The Camp-On tone is always provided for Forced Camp-On, since Warning Tone Allowed (WTA) Class of Service is a prerequisite. This tone can be a buzz for Meridian 1 proprietary telephones or a single burst of tone for analog (500/2500 type) telephones if the customer option Periodic Camp-On Tone Denied (CTD) is selected in LD 15. If the customer option Periodic Camp-On Tone Allowed (CTA) is selected in LD 15, the Camp-On Tone as defined in the Flexible Tones and Cadences (FTC) LD 56 in response to the CAMP prompt will be used. The Priority Override tone used is the same tone as used for Override; this tone is defined in response to the OVRD prompt in the FTC LD 56.

While camping on, the party attempting the Camp-On, either telephone A or E, receives ringback if the Station Camp-On (SCMP) package 121 is not equipped, or receives either ringback or busy tone, as defined by the response to the Station Camp-On Busy tone (STCB) prompt in LD 15 if the SCMP package is equipped.

Override will take place on any established call when the Flexible Feature Code (FFC) is dialed or the Override (OVR) key is pressed. That means if telephone A calls telephone B while telephone B is busy and telephone B disconnects from that call and is established on another call when telephone A activates Override, the new call will be overridden.

Example 1
Enhanced Override with an analog (500/2500 type) telephone

With automatic Forced Camp-On turned off; Response to AFCO in LD 15 was “NO”.

	ACTION	RESPONSE
1	B and C are connected in a simple call.	
2	A dials B.	A receives busy tone.
3	A performs a recall.	A receives special dial tone (SDT).
4	A dials OVRD FFC to attempt Priority Override.	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise, a conference is established between A, B, and C with Override tone given.
– or –		
	a) A dials EOVR FFC to attempt Forced Camp-On.	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise B receives Camp-On tone and A receives ringback or busy tone depending on the options equipped. A is manually forced camped on to B.
	b) B disconnects from the call.	Telephone A rings telephone B.
	– otherwise –	
	b) A performs a recall.	A receives SDT.
	c) A dials EOVR FFC to attempt Priority Override.	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise, a conference is established between A, B, and C with Override tone given.
5	If any party disconnects…	A simple two-party call is established.

With automatic Forced Camp-On turned on; response to AFCO in LD 15 was "YES".

	ACTION	RESPONSE
1	B and C are connected in a simple call.	
2	A dials B.	A attempts Forced Camp-On to B.
	a) If Forced Camp-On was successful...	A receives ringback or busy tone depending on the options equipped. A is automatically forced camped on to B.
	b) B disconnects.	A rings B.
	– otherwise –	
	a) A performs a recall and dials the OVRD or EOVR FFC to attempt Priority Override.	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise, a conference is established between A, B, and C with Override tone given.
– or –		
	a) If Forced Camp-On was unsuccessful due to Class of Service restrictions...	A receives busy tone.
	b) A performs a recall and dials OVRD or EOVR FFC to attempt Priority Override.	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise, a conference is established between A, B, and C with Override tone given.
– or –		
	a) If Forced Camp-On was unsuccessful due to other limitations, then Priority Override is also restricted.	A receives busy tone.
	b) A performs a recall and dials OVRD or EOVR FFC to attempt Priority Override.	A receives overflow (fast busy) tone.

Example 2
Enhanced Override with a Meridian 1 proprietary telephone

With automatic Forced Camp-On turned off; response to AFCO in LD 15 was "NO".

	ACTION	RESPONSE
1	B and C are connected in a simple call.	
2	E dials B.	E receives busy tone.
3	E presses OVR key to attempt Priority Override.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, a conference is established between E, B, and C with Override tone given.
– or –		
	a) E presses EOVR key to attempt Forced Camp-On.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, B receives Camp-On tone and E receives ringback or busy tone depending on the options equipped. E is manually forced camped on to B.
	b) B disconnects from the call.	Telephone E rings telephone B.
	– otherwise –	
	b) E presses EOVR key to attempt Priority Override.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, a conference is established between E, B, and C with Override tone given.
4	If any party disconnects…	A simple two-party call is established.

With automatic Forced Camp-On turned on; response to AFCO in LD 15 was "YES".

	ACTION	RESPONSE
1	B and C are connected in a simple call.	
2	E dials B.	E attempts Forced Camp-On to B.
	a) If Forced Camp-On was successful…	E receives ringback or busy tone depending on the options equipped. E is automatically forced camped on to B.
	b) B disconnects.	E rings B.
	– otherwise –	
	a) E presses OVR or EOVR key to attempt Priority Override.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, a conference is established between E, B, and C with Override tone given.
– or –		
	a) If Forced Camp-On was unsuccessful due to Class of Service restrictions…	E receives busy tone.
	b) E presses OVR or EOVR key to attempt Priority Override.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, a conference is established between E, B, and C with Override tone given.
– or –		
	a) If Forced Camp-On was unsuccessful due to other limitations, Priority Override is also restricted.	E receives busy tone.
	b) E presses OVR or EOVR key to attempt Priority Override.	A receives overflow (fast busy) tone.

Example 3
Enhanced Override from a consultation call with an analog (500/2500 type) telephone

With automatic Forced Camp-On turned off; Response to AFCO in LD 15 was "NO"; Station-to-Station Camp-On is denied or Station-to-Station Camp-On is equipped and D is a station; Multi-Party Operation active.

	ACTION	RESPONSE
1	A is connected to D and B and C are connected in a simple call.	
2	A performs a recall.	A receives special dial tone (SDT). D is held.
3	A dials B.	A receives busy tone.
4	A releases.	Treated as misoperation of call transfer.
– or –		
	a) A performs a recall and dials any control digit.	A releases from B and returns to D.
– or –		
	a) A performs a recall.	A receives control dial tone.
	b) A dials OVRD FFC to attempt Priority Override.	Conference is established between A, B, and C with override tone given.
– or –		
	a) A performs a recall.	A receives control dial tone.
	b) A dials EOVR FFC to attempt Forced Camp-On.	B receives Camp-On tone. A receives ringback or busy tone depending on the options equipped. A is manually forced camped on to B.
	– if –	
	c) A releases…	D is camped on to B.
	– if –	
	c) B disconnects.	A rings B.
	– if –	
	c) A performs a recall and dials any control digit.	A releases from B and returns to D.
	– if –	

A performs a recall and dials the POVR FFC again.	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise, a conference is established between A, B, and C with Override tone given.

With automatic Forced Camp-On turned off; response to AFCO in LD 15 was "YES"; Station-to-Station Camp-On is allowed and D is an external call; Multi-Party Operation active.

	ACTION	RESPONSE
1	A is connected to D and B and C are connected in a simple call.	
2	A performs a recall.	A receives special dial tone (SDT). D is put on hold.
3	A dials B.	B receives Camp-On tone. A receives ringback or busy tone depending on the options equipped. A is automatically forced camped on to B.
4	A releases.	D is camped on to B.
– or –		
	B disconnects	A rings B.
– or –		
	A performs a recall and dials any control digit.	A releases from B and returns to D.
– or –		
	a) A performs a recall.	A receives control dial tone.
	b) A dials OVRD or EOVR to attempt Priority Override.	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise, a conference is established between A, B, and C with Override tone given.

Example 4
Enhanced Override from a consultation call with a Meridian 1 proprietary telephone

With Automatic Forced Camp-On turned off; Response to AFCO in LD 15 was "NO"; Station-to-Station Camp-On is denied or Station-to-Station Camp-On is equipped and D is a station; Multi-Party Operation active.

	ACTION	RESPONSE
1	E is connected to D and B and C are connected in a simple call.	
2	E presses Conference or Transfer key.	E receives dial tone. D is put on hold.
3	E dials B.	E receives busy tone.
4	E releases or presses Conference or Transfer key again.	Treated as misoperation of call transfer.
– or –		
	E presses the DN key that D is held on.	E is reestablished with D.
– or –		
	E presses OVR key to attempt Priority Override.	Conference is established between E, B, and C with Override tone given.
– or –		
	a) E presses EOVR key.	B receives Camp-On tone. E receives ringback or busy tone depending on the options equipped. E is manually forced camped on to B.
	– if –	
	b) E presses Transfer key.	D is camped on to B.
	– if –	
	b) B disconnects.	E rings B.
	– if –	
	b) E releases.	Camp-On is cancelled and E must press DN key to reconnect to D.
	– if –	
	b) E presses Conference or Hold key.	Key operation is ignored.
	– if –	
	b) E presses the DN key that D is held on.	E is reestablished with D.

– if –

b) E presses EOVR key again.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, a conference is established between E, B, and C with Override tone given.

With Automatic Forced Camp-On turned off; response to AFCO in LD 15 was "YES"; Station-to-Station Camp-On is allowed and D is an external call; Multi-Party Operation active.

	ACTION	RESPONSE
1	E is connected to D and B and C are connected in a simple call.	
2	E presses Conference or Transfer key.	E receives dial tone. D on hold.
3	E dials B.	E receives ringback or busy tone depending on the options equipped. E is automatically Forced Camped on to B.
4	E presses Transfer key.	D is camped on to B.
– or –		
	B disconnects.	E rings B.
– or –		
	E releases.	Camp-On is canceled and E must press DN key to reconnect to D.
– or –		
	E presses Conference or Hold key.	Key operation is ignored.
– or –		
	E presses the DN key that D is held on.	E is reestablished with D.
– or –		
	E presses EOVR or OVR key to attempt Priority Override.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, a conference is established between E, B, and C with Override tone given.

Operation with various combinations of Forced Camp-On and Priority Override

The following tables show what happens when either Forced Camp-On or Priority Override are denied.

Forced Camp-On is denied by the new Camp-On From Another Telephone Denied (CPFD) and Camp-On To Another Telephone Denied (CPTD) Classes of Service.

Priority Override is denied for analog (500/2500 type) telephones by setting the Override Denied (OVRD) Class of Service, or for all telephones by setting their Priority Override Levels (PLEV) to 0.

Both Forced Camp-On and Priority Override are denied by the Warning Tone Denied (WTD) Class of Service, or if any of the limitations described in the Operating parameters or Feature interactions section is encountered.

The following table highlights the various combinations and the results of different actions for a simple call.

Setup						
AFCO setting in LD 15	NO	NO	NO	YES	YES	YES
Forced Camp-On Allowed	NO	NO	YES	NO	NO	YES
Priority Override Allowed	YES	NO	NO	YES	NO	NO
Action	**Result**					
A dials B B is busy	BT	BT	BT	BT	BT	BT or R
A recalls analog (500/2500 type) telephones only	SDT	SDT	SDT	SDT	SDT	SDT
A uses OVR key or OVRD FFC	POVR	O&L	O&L	POVR	O&L	BT or R
OR A uses EOVR key or FFC	BT	BT	BT or R	POVR	BT	BT or R
A uses EOVR key or FFC again	POVR	O&L	BT or R	POVR	O&L	BT or R

Legend:

BT – Busy tone returned to A.

BT or R – Busy tone or ringback returned to A; A camped on to B.

O&L – Overflow (fast busy) returned to A for 30 seconds, then A is locked out.

POVR – Priority Override is attempted.

SDT – Special dial tone is returned to A.

The following table highlights the various combinations and the results of different actions for a consultation call.

Setup						
AFCO setting in LD 15	NO	NO	NO	YES	YES	YES
Forced Camp-On Allowed	NO	NO	YES	NO	NO	YES
Priority Override Allowed	YES	NO	NO	YES	NO	NO
Action	**Result**					
A connected to D A recalls analog (500/2500 type) telephones only	SDT	SDT	SDT	SDT	SDT	SDT
A dials B D is held. B is busy.	BT	BT	BT	BT	BT	BT or R
A recalls analog (500/2500 type) telephones only	CDT	CDT	CDT	CDT	CDT	CDT
A uses OVR key or OVRD FFC	POVR	O&R	O&R	POVR	O&R	BT or R
OR A uses EOVR key or FFC	BT	BT	BT or R	POVR	BT	BT or R
A uses EOVR key or FFC again	POVR	O&R	BT or R	POVR	O&R	BT or R
A recalls analog (500/2500 type) telephones only	CDT	REC	CDT	CDT	REC	CDT
OR A presses DN key on which D is held	REC	REC	REC	REC	REC	REC

Legend:

BT – Busy tone returned to A.

BT or R – Busy tone or ringback returned to A; A camped on to B.

CDT – Control dial tone returned to A.

O&R – Overflow (fast busy) returned to A for 30 seconds, then A is reconnected to D.

POVR – Priority Override is attempted.

SDT – Special dial tone is returned to A; D is held.

If at any time invalid digits are dialed for the EOVR or OVRD FFC, overflow (fast busy) tone is returned to the telephone attempting to override. This telephone receives overflow (fast busy) tone for 30 seconds and is then locked out or reconnected to the telephone on hold. If the attempted override is made from a consultation call, the telephone may perform a recall during overflow (fast busy) tone, and return to the call being held.

Enhanced Override from a conference call with any telephone

Once a consultation conference (i.e., party D is still on hold) has been established between telephone A or E and parties B and C, any of the following may occur.

ACTION	RESPONSE
Telephone B or C disconnects.	Telephone A or E remains in simple two party consultation with remaining telephone (B or C).
– or –	
Telephone A performs a recall and dials a control digit.	Multi-Party operation for control digit is dialed.
– or –	
Telephones B and C disconnect.	Telephone A or E may automatically be returned to telephone D or may have to perform a recall, depending on Class of Service (AO6/C6A and XFA). Override tone is removed.
– or –	
Telephone A disconnects or telephone E presses Transfer or Conference key.	D is transferred into the conference with B and C. Override tone is removed.
– or –	
Telephone E disconnects.	Telephones B and D remain connected. Telephone D is treated as in the case of misoperation of call transfer. Override tone is removed.

Paging

The Meridian 1 provides switching access and trunk circuit interface to a customer-supplied speaker or radio paging equipment. Paging equipment is accessed by dial access or a Page key on Attendant Consoles. Telephones cannot be assigned a Page key and must dial access this feature.

Attendant Consoles using the Page key preempt telephones having only dial access. Telephones preempted by the attendant are disconnected and must re-access the paging trunk.

Time Forced Disconnect (TFD), X11 Release 15 and later software, provides a variable timer to force disconnect Paging trunks. The timer is defined on a route basis to limit the time a user can keep a Paging trunk seized. When the timer expires, the call is disconnected from the trunk. The trunk is disconnected when the Time Forced Disconnect (TFD) timer expires in all cases, regardless of the status of the trunk at the time. Timing starts as soon as the trunk is seized (not when the call is established), so the timer must allow some delay for connection time.

The Time Forced Disconnect timer is used on the following trunk types:

- COT Central Office
- DIC Dictation
- FEX Foreign Exchange
- PAG Paging trunks
- TIE Tie direct lines
- WAT Wide Area Telephone Service

Operating parameters

Station dial access to the Paging trunk is restricted by the Trunk Group Access Restriction (TGAR) code entered in LD 10 or LD 11.

Unique access codes are required for each Paging route.

Unique feature keys are assigned for each Paging route.

All Zone Paging is not available with Meridian 1, unless the customer-provided paging equipment is equipped with separate "all-zone" input.

The following requirements apply to the X11 Release 15 Time Forced Disconnect (TFD) feature:

— The timer can only be assigned on a route basis and not to individual trunks. All trunks in a route have the same timer value.

— After a timer value is changed, it does not take effect on a given trunk until that trunk is released and seized again.

— Changing a timer value to zero (0) effectively removes the TFD timer from all the trunks in that route.

— The range of the timer is one hour, in 30-second increments (0–3600). The TFD timer is independent of all other timers.

Trunks forced off by TFD are disconnected normally, accompanied by an error message (ERR4054) output on the system terminal. The error message identifies the Originating Terminal Number (TN), Terminating Terminal Number (TN), date, and time for the following trunk types:

— Analog trunks

— Digital Trunk Interface (DTI) trunks, and

— ISDN Integrated Service Links (ISL)/Primary Rate Interface (PRI) trunks.

Feature interactions

Call Forward All Calls

Calls that originate on a TIE trunk to a telephone that is redirected to a paging route are blocked.

Conference

Paging trunks cannot be used in a conference call.

Multi-Party Operations

Users of analog (500/2500 type) telephones cannot make a consultation call while connected to a paging trunk.

Private Line Routes

Route 31 cannot be assigned as a paging route on X11 Release 13 and earlier software.

Feature packaging

Paging is included in base X11 system software.

Feature implementation

LD 16 – Add or change a Paging trunk route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	0-99 0-31	Customer number. For Option 11C.
ROUTE	0-511 1-127	Route number. For Option 11C.
TKTP	PAG	Paging trunk route.
ICOG	OGT	Outgoing trunk.
ACOD	xxx...x	Trunk route access code (if the Directory Number Expansion package is equipped, this access code can have up to seven digits).
TARG	1-31	Trunk access restriction group number.

LD 16 – Define the timer for the Time Forced Disconnect feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	0-99 0-31	Customer number. For Option 11C.
ROUTE	0-511 0-127	Route number. For Option 11C.
CNTL	(NO) YES	Changes to controls or timers (default is NO).
TIMR	TFD xxxx	TFD timer, where: xxxx = 0-(30)-3600 seconds, in 30-second increments.

LD 14 – Add or change a Paging trunk within the Paging trunk route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PAG	Paging trunk.
TN	l s c u c u	Terminal Number. For Option 11C.
XTRK	XUT XEM	Universal Trunk Card (NT8D14), E&M Trunk Card (NT8D15). Prompted only for superloops and the first unit on the card.
CUST	0-99	Customer number.
SIGL	DX2 DX4 EAM EM4 LDR OAD	DX signaling (two-wire) – QPC71 only. DX signaling (four-wire) – QPC71 and NT8D15. E&M signaling (two-wire) – QPC71 and NT8D15. E&M signaling (four-wire) – QPC71 and NT8D15. Loop dial repeating – QPC71 and NT8D14/15. Outgoing automatic, incoming dial – QPC71, NT8D14/15.

STRO	IMM WNK DDL	Immediate start outgoing. Wink start outgoing. Delay dial outgoing.
SUPN	(NO) YES	Answer and disconnect supervision required.

LD 12 – Assign Paging key for an Attendant Console. No programming is required to allow the attendant dial access to Paging.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT 1250 2250	Console type.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx PAG yyy...y	Paging key, where: xx = key number (0-9 on M1250, 0-19 on M2250), and yy...y = access code of Paging trunk route.

LD 10 – Allow or deny dial access to Paging for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
TGAR	xx	Allow/deny access to Paging trunk.

LD 11 – Allow or deny dial access to Paging for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
TGAR	xx	Allow/deny access to Paging trunk.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	10.10B
Networking:	No

Partial Dial Timing

This feature allows a partial dial timer to be associated with a Direct Inward Dialing (DID) route. The End-of-dialing timer is used for partial dial timing. It is defined on a route basis and has a range from 128 to 32640 milliseconds, in increments of 128 milliseconds.

The partial dial timer is started each time that a digit is expected. If the timer expires before a complete DN is dialed, the call is given treatments as shown in the table below.

> ***Note:*** The Partial Dial Timing feature can be used with the End of Selection and End of Selection Busy features.

PRDL EOS	NO	YES	BSY
NO	N/A	Call ATTN	Overflow tone
YES	N/A	EOS signal Call ATTN	EOS signal Overflow tone
BSY	N/A	EOS, EOSB signals Overflow tone	EOS/EOSB signals Overflow tone

Operating parameters

The Public Exchange/Central Office must be equipped to handle the special signaling requirements associated with the Partial Dial Timing feature described above.

The Partial Dial Timing feature is not available on 1.5 Mbit digital trunks or Japanese Digital Multiplex Interface (DMI) trunks.

The Partial Dial feature is not supported by R2 Multifrequency Compelled Signaling.

Feature interactions

There are no interactions with other features.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Create or modify data for trunk routes:

Prompt	Response	Description
...		
PRDL	(NO) YES BSY	No partial dial timing on DID route, Partial dial timing is equipped using EOD, or Partial Dial timing is equipped using EOD, BSY signal is sent on time out.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	15.58F
Networking:	No

Periodic Camp-on Tone

This feature replaces the single buzz or burst of tone for Meridian 1 proprietary telephones, given to indicate a camped-on call, with periodic bursts of buzz or tone. The buzz or tone can be defined on a customer basis.

The Periodic Camp-On Tone applies to calls camped-on by an attendant in standalone and Integrated Services Digital Network (ISDN) environments, and camped-on from inquiry calls in standalone environments.

Operating parameters

The feature does not apply to group calls.

Feature interactions

Attendant Break-In
Attendant Busy Verify
Override

The Periodic Camp-On Tone has precedence over Break-In, Busy Verify, and Override intrusion tones.

Semi-Automatic Camp-On

Periodic Camp-On Tone stops when the camped-on call is recalled to the attendant.

Feature packaging

International Supplementary Features (SUPP) package 131.

Dependency:

— Flexible Tones and Cadences (FTC) package 125.

Feature implementation

A tone with a periodic cadence must be defined for the Camp-On feature. An existing periodic cadence may be chosen from the Master Cadence Table, or a new cadence may be defined specifically for the Camp-On tone.

LD 56 – Define a new cadence in the Master Cadence Table (if required).

Prompt	Response	Description
...		
TYPE	MCAD	Master Cadence data block.
WCAD	0-225	Cadence Number to be given the new definition. Cadence number 0 is reserved for continuous tone and is not changeable.
CDNC	xxxx xxxx ... xxxx	Cadence. On-off phases for Cadence (ten off-on cycles). Entries 1 through 15 are reserved for ringing cadences. When defining the cadences in MCAD, each phase is entered in 5 millisecond increments. The first number defines the length of the first on period. The second defines the length of the first off period. The third defines the length of the second on period, and so forth. The range of the first phase is 1-9999 increments. The range of the second phase is 0-9999 increments. The default is 0 0 0 0 0 0 0 0 0 0.

LD 56 – Assign a cadence, either new or existing, to the Camp-On tone.

Prompt	Response	Description
...		
TYPE	FTC	Flexible Tones and Cadence data block.
CDNC	xxxx ... xxxx	The cadence number of the existing cadence, or the cadence number given to the newly defined cadence.
...		
SCCT	(NO) YES	Software Controlled Cadences and Tones. Modification of the software controlled definitions allowed.
- CAMP		Camp-On tone.
- - TDSH	i bb c tt	Tone definition for systems equipped with Tone and Digit cards, where: i = internal (0), or external (1) source bb = burst cc = cadence, and tt = frequency. Prompts with the response i bb c tt define the internal/external source, burst, cadence and frequency/level respectively. Enter the decimal equivalent (0-15) of the TDS Hex code. The first field is usually 0. If an external source is used, the entry is 1 and the fourth field is 0-7 for the specified channel.
- - XTON	0-255	XCT tone code.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	10.10B
Networking:	No

Periodic Clearing

The Periodic Clearing Signal (PCS) is used to disconnect calls that have been answered, but are now either ringing, held (consultation hold), parked (on hold without consultation), or camped-on (in the process of being transferred to a busy extension). These calls receive PCS pulses that will serve to disconnect the call if the caller has hung up. If the caller is still waiting, the line remains connected. The Periodic Clearing feature includes a Disconnect Timer (DCTI) that indicates the time period (in seconds) before a call is disconnected. The timer can be used to disconnect a call even if the periodic clearing is disabled.

Operating parameters

This feature applies only to 2 Mbit digital incoming Public Switched Telephone Network (PSTN) and Direct Inward Dialing (DID) calls.

Feature interactions

AC15 Recall: Timed Reminder Recall

When the Periodic Clearing feature is active the Disconnect timer will interfere with the AC15 recall timer. The Disconnect timer is activated on a TIE trunk or an incoming Direct Inward Dialing (DID) or Central Office (CO) trunk which is connected to the AC15 TIE trunk. If the Disconnect timer expires first the AC15 recall is cancelled and the trunk is disconnected. This is the case with a call which has been established with a TIE trunk or an incoming call on a DID or CO trunk that has been extended over an AC15 TIE trunk with the timed recall activated.

Generic XFCOT Software Support

Periodic Clearing is the sending of periodic signal from the Meridian 1 to a Central Office when an incoming call has been answered but is not in an established state (for instance, ringing, held, or parked). The connection is disconnected if the originator goes on-hook.

The Periodic Clearing condition is timed by the disconnect timer (DCTI) to prevent this situation from lasting for an extended time. When the DCTI timer expires the trunk is disconnected.

The Disconnect Timer can be used without having the feature Periodic Clearing configured particularly when the Central Office trunk has no disconnect supervision. It can be disabled by setting the DCTI to 0 in LD 16.

A loop start trunk can be marked as disconnect supervised. When it has a class of service providing disconnect supervision, in Periodic Clearing condition the trunk is disconnected when the calling station releases the call.

Feature packaging

International Supplementary Features (SUPP) is package 131.

Feature implementation

LD 16 – Create or modify data for trunk routes.

Prompt	Response	Description
...		
PECL	(NO) YES	(Do not send) send Periodic Clearing signal.

Feature operations

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	14.47E
Networking:	No

Periodic Clearing Enhancement

This enhancement permits a Meridian 1 to send a Periodic Clearing Signal (PCS) and/or start the Disconnect Timer (DCTI) on a TIE or TIE AUTO line, when a call is answered but not established to a station and there is more than one analog (500/2500 type) telephone involved in the call.

A Meridian 1 can perform the following:

— Receive a PCS on a TIE trunk, then retransmit it to another TIE trunk or incoming 2 Mbps digital or analog Public Exchange/Central Office (CO) or Direct Inward Dialing (DID) trunk, and

— Start the DCTI for incoming Central Office or DID trunks.

Operating parameters

There are no feature requirements.

Feature interactions

Called Party Disconnect Control
Toll Operator Break-in

The Called Party Disconnect Control and Toll Operator Break-in can exist on the same system and function on the same routes, but are not to be used in conjunction with Periodic Clearing.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Create or modify data for trunk routes:

Prompt	Response	Description
...		
PECL	(NO) YES	(Do not send) send Periodic Clearing signal.
DCTI	(0)-511	The time (in seconds) an extension is allowed to ring or be on hold before the trunk is disconnected. 0 specifies disconnection will not occur.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	18.20H
Networking:	No

Periodic Clearing on RAN, Meridian Mail, ACD and Music

This enhancement to the Periodic Clearing feature allows the periodic clearing signal to be sent in situations where an incoming call has been answered and connected to Meridian Mail, Automatic Call Distribution (ACD) queue, music, or a recorded announcement (including when the call has been forwarded to a pager, connected to Recorded Announcement (RAN), and placed in the pager queue). The periodic clearing signal is sent on incoming calls over Public Exchange/Central Office, Direct Inward Dialing (DID), TIE, 2 Mbps Primary Rate Interface (PRI2) TIE, and Integrated Services Digital Network Signaling Link (ISL) TIE trunks.

Operating parameters

There are no feature requirements.

Feature interactions

Called Party Disconnect Control

This feature enhancement is not supported if used together with Toll Operator Break-In.

Centrex Switchhook flash

This feature enhancement is not supported if used together with Centrex Switchhook flash.

Integrated Services Digital Network (ISDN) Basic Rate Interface

This feature enhancement is not supported on ISDN Basic Rate Interface.

MFC and MFE signaling

This feature enhancement is not supported if used on MFC and MFE signaling trunks.

Toll Operator Break-In

This feature enhancement is not supported if used together with Toll Operator Break-In.

Feature packaging

International Supplementary Features (SUPP) package 131; and Network Attendant Service (NAS) package 159.

Feature implementation

LD 15 – Allow or deny Periodic Clearing on Meridian Mail for a customer.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
...		
OPT	(PCMD) PCMA	Deny (the default) or allow Periodic Clearing on Meridian Mail.

LD 23 – Configure Periodic Clearing on Meridian Mail.

Prompt	Response	Description
...		
PCMM	(NO) YES	Deny (the default) or allow Periodic Clearing on Meridian Mail. Prompted only if OPT = PCMA in LD 15.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	10.10C
Networking:	No

Periodic Pulse Metering

The Periodic Pulse Metering (PPM) feature allows the user of each station within a Meridian 1 to keep an accurate record of Public Switched Telephone Network (PSTN) and Direct Outward Dialing (DOD) calls for billing or administration. The PPM feature:

— Detects rapid PPM (the system will be able to detect and count at least three pulses per second).

— Records the accumulated PPM count for each call on the Call Detail Reporting (CDR) if equipped.

— Calculates and records the total charge for each call based on the assigned unit and the total number of received pulses for the call.

— Allows the attendant to mark a specified call in order to read out the number of accumulated PPM counts against this call.

— Allows the customer to specify a particular schedule for printing the MR reports.

— Supports Call Detail recording (CDR) on multiple call transfer for outgoing PPM calls.

Operating parameters

A Periodic Pulse Meter can count to a maximum of 32,767 pulses. When this limit is exceeded, an indication of overflow is provided.

To access message registration data, an SL-1 telephone must have a digit display. Hence, MR data access from the M2009, M2112, and M2018 is not supported.

The reading and changing of Periodic Pulse Meters from the M3000 telephone is not supported.

PPM is not supported by the 1.5 Mbit Digital Trunk Interface (DTI).

Feature interactions

AC15 Recall: Transfer from Norstar

If party Z (on Norstar) calls party X and transfers the call to party Y, if party X is an outgoing trunk with PPM or Advice of Charge on the Meridian 1, the call is charged against the AC15 trunk route's meter until the transfer is completed. When party Z completes the transfer in ringing status, the charges still accumulate in the AC15 trunk route's meter. If the call is in established status, the charges accumulate against party Y, if party Y has a meter, or otherwise against the customer meter.

Advice of Charge for EuroISDN

Advice of Charge has the following interactions with the Periodic Pulse Metering (PPM): recording of accumulated call charging information for each call on the CDR record, calculating the total charge for each call based on the assigned unit cost and the accumulated information received from the network, allowing the attendant to read the number of call charge units on a per call basis and allowing a set with a MRK key to access Message Registration information.

Attendant Administration

Attendant Administration does not support the PPM feature.

Call Detail Recording

If both the Call Detail Recording (CDR) and Meter Registration feature are equipped for a customer, the PPM pulse counts for metered calls over trunks for which the CDR feature is enabled are recorded on the CDR record along with the standard CDR information. If the charge option is allowed, the charge for the call is calculated and recorded on the CDR. If the charge option is disabled, zeros are printed in the charge field on the CDR. As a customer option, the CDR records can be printed onto a teletype terminal or tape unit.

Call Forward All Calls
Call Forward No Answer
Hunting

Metered calls transferred or extended from one station to another using the Call Forward All Calls, Call Forward No Answer, or Hunting feature are charged against the last station at which the call is answered as the controlling station releases. The last party to forward a call onto a metered PPM trunk is charged.

Call Park

When a metered call is parked from one station to another, the controlling station is charged until the call is answered.

Call Pickup

Metered calls transferred or extended from one station and answered at another station using the Call Pickup feature are charged against the station where the call is picked up as the controlling party disconnects.

Call Transfer

If the user of a station which is connected to a metered trunk transfers an internal call to another internal station while the dialed station is still ringing, the PPM pulse count is accumulated against the transferring station until the call is answered by the dialed party, or abandoned by the dialing party. When the call is answered, the pulses are counted against the station to which the call has been transferred.

If the station user transfers the call after consulting with the dialed station user, then the PPM pulses are counted against the controlling station until the call is transferred. When the call is transferred, the PPM pulses are counted against the station to which the call has been transferred. If the transferred call is redirected using any of the call redirection features such as Call Forward or Hunting, the call is charged against the transferring station until the call is transferred. The pulses are then counted against the answering station. This method ensures that PPM meters are charged in a manner consistent with the printing of CDR records.

Camp-On

Metered calls camped-on to a busy station by an attendant are charged against the attendant until the call is answered and the attendant releases.

Conference - Attendant

If an attendant establishes a conference which includes one or more metered trunks, and the attendant first dials a metered trunk as a source, the PPM pulses are counted and accumulated against the attendant. If the attendant continues to hold the conference at the console, the pulses continue to accumulate against the attendant. If the attendant releases the conference from the console, the pulses are accumulated against the station that has been in conference the longest. If the attendant first dials an internal station or a TIE trunk, any connection established thereafter is charged against this station or trunk.

Conference - Three-party/Six-party

Whenever a PPM trunk is added to a conference, a CDR Start record is generated, if CDR is equipped on the trunk. The PPM pulse counts from the trunk are accumulated against the party who initiated the call. If a party who adds a PPM trunk to the conference disconnects while the conference is still in progress, read requests are sent to the PPM trunk to read the residual count. Then, the on-board counter is cleared, the residual count is added to the temporary meter, and the contents of the temporary meter are added to the terminal meter. A CDR Transfer (X) record is then printed against this party, and the temporary meter is cleared. The party that is charged is the one that has been in conference the longest. When a trunk with disconnect supervision disconnects, a CDR End record is immediately printed. For trunks that do not provide a disconnect signal, their CDR records are not printed until the last party disconnects from the conference.

Consultation calls

If a user establishes a consultation call including one or more metered trunks, all the associated pulses are counted against the controlling station until the call is transferred.

Digital Trunk Interface (DTI) – Commonwealth of Independent States (CIS)

Periodic Pulse Metering is not supported by CIS DTI.

Italian Central Office Special Services

Periodic Pulse Metering pulses are received from the Central Office according to the charge of the accessed service, and are collected and stored as per normal procedures.

Italian Periodic Pulse Metering

This feature now allows PPM pulses to be counted on Italian DTI2 trunks. The Italian DTI2 option default is set to NA (that is, not active when software prior to the introduction of this feature is upgraded). Existing operation thus continues unaffected by the new feature.

Recall to Same Attendant

Meter recalls are returned to the same attendant whether Recall to Same Attendant is allowed or not. If Return to Same Attendant with Queuing on Busy (RSAQ) is selected as an option, the recalls are queued to a specified attendant.

Tandem Switching

If an incoming TIE trunk is connected to a PPM trunk, the pulses are counted against the access code of the TIE route.

Virtual Network Service

Periodic Pulse Metering is supported on the Virtual Network Service Bearer trunks only.

1.5 Mbps Digital Trunk Interface

PPM is not supported by 1.5 Mbps DTI.

2 Mbps Digital Trunk Interface

PPM operates the same for 2 Mbps DTI as for analog trunks.

Feature packaging

Periodic Pulse Metering/Message Registration (MR) package 101.

Feature implementation

LD 17 – For Release 20.0x and later only, select PPM functionality in the Configuration Record.

Prompt	Response	Description
REQ	CHG	Change
TYPE	PARM	Release 19 gate opener.
...		
MTRO	PPM	Periodic Pulse Metering meter option

LD 12 – Create or modify the data blocks for Attendant Consoles.

Prompt	Response	Description
...		
KEY	xx MTR	Add a meter key.

LD 15 – Assign Meter Incoming Call Indicator.

Prompt	Response	Description
...		
ICI	xx MTR	xx is the selected key/lamp number.

LD 16 – Create or modify data for each DID trunk route data block to have or deny MFC Signaling option.

Prompt	Response	Description
...		
CDR	(NO) YES	Call Detail Recording for the trunk route.
MR	PPM	Message Registration Buffered PPM signal to be counted on this route.

LD 14 – Trunk Data Blocks must be created or modified.

Prompt	Response	Description
...		
SIGL	GRD LOP	Signaling start arrangement, Ground or Loop.
Prior to Release 20.0X program the following:		
CLS	PIP PSP LNT	Polarity Insensitive Pack Polarity Sensitive Pack, or Unsupervised loop start trunk.
For Release 20.0x and later program the following:		
SUPN	YES (NO)	Trunk Supervision required (not required)
STYP	PSP (PIP)	Polarity sensitive packs. Polarity insensitive packs.

Feature operation

If the attendant desires billing information immediately upon the completion of a long distance call, the call must be flagged by the attendant as a metered call. When a metered call is terminated or modified, the same attendant is recalled and the calculated call charge or PPM count for this call is displayed on the console. If the call is transferred, a Meter Recall will be routed to the attendant for each portion of the trunk connection.

The following keys are added the Attendant Console for this feature:

— The **MTR** key and lamp that can be assigned at any position on the flexible feature key strip on the Attendant Console, and

— The **Meter Recall ICI** key and lamp that can be assigned at any ICI position on the Attendant Console.

Marking a Call as Metered

The attendant can request the call charge or PPM count on any outgoing PPM call by pressing the **MTR** key after the PPM call has been made. When the **MTR** key is pressed, the meter lamp is lit and all the metered outgoing PPM trunks connected to the active console loop (for example, as in a conference) are marked as metered. Additional PPM trunks added to the conference hereafter are marked as metered automatically. Metering a non-PPM call is ignored.

To cancel the metered flag re-press the **MTR** key.

Meter Recall

When a metered call is modified or disconnected, a meter recall is presented to the attendant.

1 The meter recall ICI lamp comes on.

2 The Source side of an idle loop is lit.

3 The Destination lamp remains off.

4 The following information appears on the display of the Attendant Consoles:

  a The DN of the station or Access Code of the TIE trunk on which the external call was placed is shown on the left-hand portion of the digit display.

  b If the option "charge to Attendant Console" is selected, the call charge is calculated by multiplying the PPM count in the temporary meter for this call by the customer assigned unit cost. The call charge is then shown on the right-hand portion of the digit display. If an overflow occurs when the charge is calculated, an overflow indication is given to the attendant – "DN-32767".

  c If the option "charge to Attendant Console" is disabled, the PPM count in the temporary meter is shown on the right-hand portion of the digit display.

If the attendant who originated the metered call is in Position Busy, the meter recall is presented to the next idle Attendant Console. It is possible for an Attendant Console unequipped with a **MTR** key to receive a meter call. If all attendants are in Night Service or Position Busy, the recall is saved in the attendant queue until one of the attendants becomes idle.

An attendant answers the meter recall by pressing the **Loop** key, and releases the Call by pressing the **Rls** key or another **Loop** key.

Introduced in X11 Release:	20.0x
Networking:	No

Phantom Terminal Numbers

The Phantom Terminal Numbers (PHTN) feature permits Meridian 1 users to define and configure Terminal Numbers (TNs) with no associated physical hardware. Normally, a TN with no associated hardware is disabled.

With Phantom TNs configured, users can define phantom Directory Numbers (DNs) as well. This feature, in conjunction with Call Forward All Calls (CFW) and Remote Call Forward (RCFW), allows a call to a phantom DN to be redirected to an existing telephone.

Phantom TNs can use loops 0-159 for all machine types except the Option 11. Phantom TNs on Option 11 telephones are restricted to card slots 41-60 (which convert to Superloops 64-80).

> *Note:* The Phantom Terminal Numbers feature is not to be used for predictive dialing applications. For information on configuring Phantom TNs for predictive dialing, please refer to the Predictive Dialing feature module in this guide.

Operating parameters

Phantom TNs can only have Single Appearance DNs.

All DNs configured on Phantom TNs must conform to the current customer-defined dialing plan.

LD 25 (Move Data Blocks) is not supported between phantom and nonphantom loops; however, it is supported between phantom loops.

Only analog (500/2500 type) telephones support Phantom TNs.

Model telephones (such as TN 500 M) are not supported.

A Phantom TN requires one of the phantom terminal loop types shown in Table 117.

Table 117
Supported phantom terminal loop types

Mnemonic	Description
TERM	Single (1) density terminal loop, configured in LD 17.
TERD	Double (2) density terminal loop, configured in LD 17.
TERQ	Quadruple (4) density terminal loop, configured in LD 17.
SUPL	Superloop (8) density terminal loop, configured in LD 97.

Feature interactions

Attendant Administration

This feature is not supported. Phantom DNs cannot be configured on a nonphantom TN.

Attendant Blocking of Directory Number

DNs on Phantom TNs will not be overridden by the Attendant Blocking of DN feature.

Automatic Call Distribution

Phantom TNs cannot be configured as Automatic Call Distribution (ACD) agents.

Call Detail Recording

Call Detail Recording records interact with a Phantom TN exactly the same as with an existing TN with its CFW feature turned on.

Call Forward All Calls

Call Forward All Calls is used in conjunction with RCFW to redirect incoming calls to a Phantom TN/DN to a valid DN.

Call Forward and Busy Status

Attempting to define a BFS key for a Phantom TN results in an error message at the beginning of the phantom loop.

Call Forward, Internal Calls

Internal Call Forward cannot be enabled on a phantom TN.

Call Forward/Hunt Override Via Flexible Feature Code

Phantom Terminal Numbers are not overridden by the Call Forward/Hunt Override Via FFC feature. If Call Forward/Hunt Override Via FFC is used against a phantom TN the call will be canceled and overflow tone will be given.

Call Forward, Remote (Attendant and Network Wide)

A Phantom TN does not physically exist, but can be configured with limited hardware associated with it (that is, no sets or line cards); however, all required data blocks are configured.

The Phantom TN feature uses the RCFW feature to configure and activate/deactivate the CFW DN on the Phantom TNs.

As the data blocks associated with Phantom TNs match those of standard analog (500/2500 type) telephones configured within the Meridian 1, the operation of the RCFA and RCFD features on Phantom TNs is applicable to the RCFW feature. As such, the set-based local and network RCFW features can be used to configure and activate/deactivate the CFW DN of Phantom TNs.

The Phantom TN feature uses a Default Call Forward (DCFW) DN. If call forward is not active on the Phantom TN, all calls to the Phantom TN DN are routed to the DCFW DN.

The Phantom TN feature modifies the set-based RCFW feature so that if CFW is not active on the Phantom TN, and the CFW DN entered in the RCFV operation matches the DCFW DN, confirmation tone is returned to the RCFV user; if the CFW DN entered does not match the CDFW DN, overflow is returned.

This change to the set-based RCFV operation is applicable to the network RCFV operation. The operation of this feature network wide requires no changes to the ISDN message passing for the set-based network RCFV operation.

There is no Attendant RCFW operation which interacts with the DCFW DN of Phantom TNs.

Hot Line

Hot Line does not support Phantom TNs.

Call Forward, Internal
Manual Line Service
Multiple Appearance
Multiple Appearance Directory Number Redirection Prime
Station Category Index

These features cannot be enabled on a Phantom TN.

DPNSS1 Diversion

If an incoming call to a Phantom TN contains a DIVERSION BY-PASS REQUEST, Call Forward All Calls applies.

Meridian Link

Phantom TNs cannot be used for origination and termination of calls. AST Class of Service is not allowed on Phantom TNs. With the Telelink Mobility Switch feature, a separate type of TN can be used by Meridian Link AST applications.

Meridian Mail

Phantom DNs are treated like other DNs; a phantom DN can have a Meridian Mail box.

Network Ring Again

The Network Ring Again (NRAG) feature is supported for a Phantom TN with Default Call Forward (DCFW) to an internal set. When the called party becomes idle, the originating caller receives a "set-free" notification. The originating party then presses the Ring Again key, and the DN of the Phantom TN is dialed.

Network Ring Again is not supported for Second Level Default Call Forward or Default Call Forward to an external set.

Override

Call Forward cannot be overridden on phantom terminal numbers. The overflow tone occurs if an Override is attempted

Recorded Announcement for Calls Diverted to External Trunks

If a Phantom TN is forwarded to an external outgoing CO route and the Recorded Announcement for Calls Diverted to External Trunks feature is configured for this route, the calling party that is forwarded due to the Phantom TN feature will be provided with a recorded announcement.

Remote Call Forward

If Remote Call Forward is to be used in conjunction with Phantom TNs, then the Phantom TNs must be configured with the Call Forward All Calls (CFW) feature.

Ring Again on No Answer

Although Ring Again on No Answer can be applied to a phantom DN, it is not recommended. Because a phantom DN cannot be active or busy, the caller is not notified when the phantom DN's forward DN does not answer.

Secretarial Filtering

If a Phantom TN is call forwarded to an existing telephone, and that telephone is used to call the DN on the Phantom TN, the call receives DCFW treatment.

Set-Based Administration Enhancements

Set-Based Administration supports making changes to Phantom TNs with the exception of changing Hunt DNs, since Phantom TNs cannot have Hunt DNs.

Feature packaging

The Phantom Terminal Numbers (PHTN) feature is available as package 254.

Using Remote Call Forwarding (RCFW) with Phantom TNs requires Flexible Feature Codes (FFC) package 139.

Feature implementation

LD 17 – Configure a phantom loop.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CEQU	Release 19 gate opener.
- TERM	N0-N159	Single density local terminal loop; precede loop number with "N" to create a phantom loop; precede with an "X" to remove a terminal loop.
- TERD	N0-N159	Double density local terminal loop; precede loop number with "N" to create a phantom loop; precede with an "X" to remove a terminal loop.
- TERQ	N0-N159	Quadruple density local terminal loop; precede loop number with "N" to create a phantom loop; precede with an "X" to remove a terminal loop.

LD 97 – Configure a phantom Superloop.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	SUPL	Superloop data.
SUPL	N0-N156 N64-N80	Local Superloop in multiples of four; for Option 11 systems, range is 64-80 in multiples of four; precede loop number with "N" to create a phantom loop; precede with an "X" to remove a terminal loop. ***Note:*** Phantom TNs can use loops 0-159 for all machine types except the Option 11. Phantom TNs on Option 11 telephones are restricted to card slots 41-60 (which convert to superloops 64-80).

LD 10 – Define a TN for the phantom loop.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	500	Telephone type.
TN	l s c u	Terminal number (loop, shelf, card, and unit); if the loop is a phantom loop, "PHANTOM" is echoed to the technician.
	c u	For Option 11C.
DN	xxx...x	Directory Number; must be a Single Appearance DN
CLS	aaaa	Class of Service options, which cannot include AGTA, CCSA, MNL, or LPA.
FTR	DCFW ll xxx...x	Default DCFW length (ll) and default CFW DN xxx...x (up to 23 digits).

Feature operation

Call forwarding occurs as described below, with the exception of the situations described in "Feature interactions" on page 2294.

1 A call is directed to a phantom DN.

2 If the phantom DN is Call Forward Activated, the call is directed to its CFW DN.

3 If the phantom DN is Call Forward Deactivated, the call is directed to its Default CFW DN.

Position Busy with Call on Hold

This feature prevents an attendant, having a call on a Loop key on hold or having either the source or destination of an active Loop key being excluded from a call, from going into Position Busy.

Operating parameters

There are no feature requirements.

Feature interactions

Attendant Forward No Answer

If an attendant with a call on hold does not answer an Attendant Forward No Answer call within a customer-defined time, the console is not placed in Position Busy.

Scheduled Access Restriction

If an attendant in a Scheduled Access Restriction group has a call on hold, the attendant is not placed in Position Busy when the group enters an off-hour period.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 15 – Configure Position Busy with Call on Hold.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
...		
- OPT	(BOHA) BOHD	Position Busy with Calls on Hold (allowed) denied.

Feature operation

With BOHA configured in LD 15, normal operation is not changed when an attendant with a call on hold presses the **POS BUSY** key. The attendant goes into Position Busy.

With BOHD configured in LD 15, when an attendant with a call on hold presses the **POS BUSY** key the system will react as if nothing has happened.

In addition, if the attendant with a call on hold presses the **POS BUSY** key, the system remains in day service if supposed to go in Night Service.

Predictive Dialing

With Predictive Dialing, the process of making outgoing calls to customers is automated for Automatic Call Distribution (ACD) agents. Host applications can request the Meridian 1 to make calls using autodialers or phantom TNs. When a call is answered, the application sends a request to the switch to transfer the call to a live agent. The call needs to be transferred before, or while, the customer starts speaking in order to prevent customers from abandoning the call if they think no one has called them. This transfer was previously performed by Meridian Link in two steps by sending two separate Application Module Link (AML) messages to initiate and then complete the transfer. This operation takes a minimum of 400 to 450 milliseconds for the Meridian 1 to process.

The Fast Transfer development in X11 Release 21 allows applications residing on the Application Module (AM) or host computers to transfer a call in one step, a blind transfer, by sending only one AML message (Fast Transfer) to the switch, thereby saving approximately 200 to 250 milliseconds of transfer time. This Fast Transfer feature is useful for predictive applications to make outbound calls and then quickly transfer them once the customer has answered (i.e., live voice has been detected). Fast Transfer can also be used in a non-predictive dialing environment. Applications that want to perform a blind transfer can now execute it more quickly.

The Predictive Dialing feature enables applications residing on the AM or host computers to send a combined Make Call and Transfer request on behalf of an autodialer or Phantom TN. As soon as live voice is detected by third-party equipment, or notification is sent to the switch indicating the call has been answered (e.g., answer supervision), the application can send the Fast Transfer request to the switch immediately transferring the call to an ACD agent.

Operating parameters

When Phantom TNs/DNs are used to originate calls as part of a predictive dialing operation, the Option 11 will not be supported.

Attendant Consoles, and Basic Rate Interface sets cannot initiate Fast Transfer or predictive calls.

The Meridian 1 does not support live voice answer detection. Live voice answer detection is currently achieved through third-party vendor equipment.

If phantom TNs/DNs are used, this development only supports calls and Fast Transfers originated by phantom TNs/DNs which are defined as Associate set (AST) Meridian 1 proprietary telephones on a phantom loop.

Data calls are not supported.

For outbound trunk calls, if no third-party equipment is used to detect live voice answer, the switch will have to depend on receiving answer supervision before transferring the call to the target DN.

If voice detection is used, the application will not be able to Fast Transfer the call before the call is established (i.e., answer notification is received).

The application will not be able to complete the transfer when Fast Transferring over a trunk.

Not all analog trunks support answer supervision. All digital trunks do not provide answer supervision. For trunks that do not support answer supervision, the End-of-Dialing (EOD) timer will be used to trigger the transfer.

Receiving answer supervision depends on the accuracy of signals returned by the external network. Answer supervision may be received before an EOD timeout, fake answer supervision may also be received due to an EOD timeout, and a pseudo answer supervision may be received if the far-end has an EOD timeout even though the local switch has answer supervision configured.

The AML requires an Enhanced Serial Data Interface (ESDI) card or Multi-purpose Serial Data Link (MSDL) card (NT6D80AA) on the switch. If an Option 11 is used, a Serial Data Interface/D-Channel (SDI/DCH) card (NTAK02AA) is required to configure the ESDI port.

The AML connection requires an RS232 cable.

Meridian Link software is required for host applications to utilize this feature.

Feature interactions

Call Hold, Deluxe
Call Hold, Permanent

If an established call is put on hold by the set initiating the Fast Transfer, the switch will not be able to transfer the call. The switch can only transfer a call if it is in the established state.

Call Transfer by Meridian 1 proprietary telephone

The application sends the Fast Transfer request on behalf of a Meridian 1 proprietary telephone, and then the switch initiates and completes the transfer immediately which is similar to a normal call transfer from a Meridian 1 proprietary telephone.

In a Predictive Dialing scenario where the autodialer (originating DN) is a Meridian 1 proprietary telephone, the Make Call message sent by the application to the switch to make a call on behalf of the Meridian 1 proprietary telephone, and then the call transfer call, will interact with the Meridian 1 proprietary telephone Call Transfer feature. The autodialer is configured with Class of Service TRN so that the switch can transfer the call to the target destination.

Call Transfer by Analog (500/2500 type) Telephone

The application sends the Fast Transfer request on behalf of an analog (500/2500 type) telephone. The switch will then initiate and complete the transfer in one step.

In a predictive dialing scenario, the application will send the Make Call request on behalf of the autodialer (analog (500/2500 type) telephone) to have the switch make the call, and then transfer the call when the switch receives the Fast Transfer message. The autodialer needs to be configured with Classes of Service Dial Pulse (DIP) and Transfer Allowed (XFA) for 500 sets, or with Classes of Service Digitone (DTN) and XFA for 2500 sets.

Command and Status Link

The Command and Status Link, also known as the AML, is the link on which the messages for the Predictive Dialing feature flow between the switch and an Application Module. The CON/FastTransfer is an AML message.

Trunks

Only certain trunks will support answer supervision. The End-of-Dialing timer will be used for trunks that do not support answer supervision.

Feature packaging

There are no new software packages required for the Predictive Dialing feature. However, the following packages are required to utilize the feature:

— Application Module Link (IAP3P) package 153, and

— Meridian Link Module (MLM) package 209 if the Meridian Link Module is involved.

Feature implementation

This feature does not require any changes to the overlays. The following illustrates the configuration requirements to set up this feature. Most of these requirements are used by existing Meridian Link and Application Module applications.

LD 17 – Configure the ESDI port to the Meridian Link Module.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Release 19 gate opener.
- CTYP	ESDI	Card Type. ESDI card.

- DNUM	x	Device number is x.
- DES	NEWTTY	Description of this I/O device.
- BPS	19200	Baud rate is 19,200 bits per second.
- CLOK	INT	Internal clocking.
- IADR	3	HDLC protocol individual address.
- RADR	1	HDLC protocol remote address.
TYPE	PARM	Release 19 gate opener.
...		
- CSQI	(20)	Maximum call registers for Command and Status Link (CSL) input queues (use the default, unless the system requires otherwise).
- CSQO	(20)	Maximum call registers for CSL output queues (use the default, unless the system requires otherwise).
TYPE	VAS	Release 19 gate opener.
...		
- VSID	y	Server ID y.
- AML	x	Port used by AML defined earlier in this overlay.
- - SECU	YES	Security on for Meridian Link.
- - INTL	x	Length of time interval (five-second increments) (e.g., 2).
- - MCNT	x	Threshold for number of messages per time interval (e.g., 100).
- - CONF	DIR	Direct link configuration.

LD 17 – Configure the MSDL port to the Meridian Link Module.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Release 19 gate opener.
...		
- CTYP	MSDL	Card Type. MSDL card.
- DNUM	y	Device number is y. Refers to the device number on the MSDL card.
- DES	MERIDIAN_LINK	Description of this I/O device.
- BPS	19200	Baud rate is 19,200 bits per second.
- PARM	RS232 DCE	Parameters for interface and transmission mode. DTE/DCE setting.
- IADR	3	HDLC protocol individual address.
- RADR	1	HDLC protocol remote address.
TYPE	PARM	Release 19 gate opener.
...		
- CSQI	(20)	Maximum call registers for CSL input queues (use the default, unless the system requires otherwise).
- CSQO	(20)	Maximum call registers for CSL output queues (use the default, unless the system requires otherwise).
TYPE	VAS	Release 19 gate opener.
...		
- VSID	y	Server ID y.
- AML	x	Port used by AML x, defined earlier in this overlay.
- - SECU	YES	Security on for Meridian Link.
- - INTL	x	Length of time interval (five-second increments) (e.g., 2).

- - MCNT	x	Threshold for number of messages per time interval (e.g., 100).
- - CONF	DIR	Direct link configuration.

LD 10 – Configure non-ACD analog (500/2500 type) telephones as autodialers.

Prompt	**Response**	**Description**
REQ	CHG	Change.
TYPE	500	Telephone type.
...		
TN	l s c u c u	Terminal Number. Terminal Number for the Option 11.
...		
CUST	0-99 0-31	Customer number. For Option 11C.
...		
DN	x...x	Internal Directory Number.
AST	YES	Associate set assignment. The internal DN is an AST.
CLS	XFA	Transfer allowed.
CLS	DIP	Dial Pulse Class of Service for 500 sets (use DTN for 2500 sets).

LD 11 – Configure non-ACD Meridian 1 proprietary telephones as autodialers.

Prompt	Response	Description
REQ	NEW	New.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. Terminal Number for the Option 11.
...		
CUST	0-99 0-31	Customer number. For Option 11C.
...		
KLS	1-7	Number of key lamp strips, typically one.
...		
AST	xx yy	Key number for Associate set DN assignment.
...		
KEY	xx SCR yyyy	Key number, Single Call Ringing, DN.
KEY	xx TRN	Key number, Call Transfer.
KEY	xx AO6	Key number, six-party conference.
KEY	xx SCR yyyy	Key number, Single Call Ringing, second DN.
CLS	xx RLS	Key number, Release.

LD 23 – Configure ACD groups.

Prompt	Response	Description
REQ	NEW	New.
TYPE	ACD	Automatic Call Distribution data block.
CUST	0-99 0-31	Customer number. For Option 11C.
ACDN	xxxx	ACD Directory Number.
...		
ISAP	YES	Integrated Services Application Protocol. ACD DN uses Meridian Link (ISDN/AP) messaging.
- VSID	0-15	Value Added Server ID. This Server ID used for Meridian Link messaging must match the VSID defined in LD 17.

LD 10 – Configure ACD analog (500/2500 type) telephones as autodialers.

Prompt	Response	Description
REQ	NEW	New.
TYPE	500	Telephone type.
...		
TN	l s c u c u	Terminal Number. Terminal Number for the Option 11.
CUST	0-99 0-31	Customer number. For Option 11C.
...		
DN	x...x	Internal Directory Number.
AST	YES	Associate set assignment. The internal DN is an AST.
...		
CLS	AGTA	ACD agent allowed Class of Service.

CLS	DIP	Dial Pulse Class of Service for 500 sets (use DTN for 2500 sets).
...		
AACD	YES	ACD telephone is an Associate set.
FTR	ACD xxxx yyyy	ACD DN and the ACD position ID.

LD 11 – Configure ACD Meridian 1 proprietary telephones as autodialers.

Prompt	**Response**	**Description**
REQ	NEW	New.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. Terminal Number for the Option 11.
...		
CUST	0-99 0-31	Customer number. For Option 11C.
...		
KLS	1-7	Number of key lamp strips, typically one.
...		
AST	xx yy	Key numbers for Associate set DN assignment.
...		
KEY	0 ACD xxxx yyyy	Key 0, ACD, ACD DN, and agent's ID.
KEY	xx MSB	Key number, Make Set Busy.
KEY	xx NRD	Key number, Not Ready.
KEY	xx TRN	Key number, Call Transfer.
KEY	xx AO6	Key number, six-party conference.

KEY	xx SCR yyyy	Key number, Single Call Ringing, second DN.
CLS	xx RLS	Key number, Release.

LD 23 – Configure a Control DN (CDN – default mode). If the application wants to transfer a call to a target CDN, a CDN must be configured. CDNs can be in default or controlled mode.

Prompt	**Response**	**Description**
REQ	NEW	New.
TYPE	CDN	Control Directory Number data block.
CUST	0-99 0-31	Customer number. For Option 11C.
CDN	xxxx	DN of the Control DN (counts as an ACD DN).
...		
DFDN	xxx...x	Default destination ACD DN.
CEIL	0-(2047)	CDN ceiling value. CEIL limits the number of unanswered calls a CDN can have at its default ACD DN at a time. Enter the maximum value (the default).
...		
RPRT	YES	Report Control.
CNTL	NO	NO sends CDN calls to the Default ACD DN.

LD 23 – Configure a Control DN (CDN – controlled mode).When a CDN is in controlled mode, the application can have control of the call once it enters the CDN.

Prompt	Response	Description
REQ	NEW	New.
TYPE	CDN	Control Directory Number data block.
CUST	0-99 0-31	Customer number. For Option 11C.
CDN	xxxx	DN of the Control DN (counts as an ACD DN).
...		
DFDN	xxx...x	Default destination ACD DN.
CEIL	0-(2047)	CDN ceiling value. CEIL limits the number of unanswered calls a CDN can have at its default ACD DN at a time. Enter the maximum value (the default).
...		
RPRT	YES	Report Control.
CNTL	YES	Control DN is in control (the default).
VSID	0-15	Value Added Server ID. Server ID used for Meridian Link messaging (defined in LD 17).
HSID	0-15	Host Link ID used when Customer Controlled Routing and Meridian Link applications are both running.

LD 14 – Define answer supervision for trunks. If the application wants to transfer outgoing calls based on answer supervision, answer supervision must be configured. If answer supervision is not configured, the End-of-Dialing timer will be used as a trigger for the Meridian 1 to transfer the call.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaa	Trunk type where: aaa = CAA, CAM, COT, CSA, DID, FEX, FGDT, IDA, TIE, or WAT.
TN	l s c u c u	Terminal Number. Terminal Number for the Option 11.
...		
SUPN	YES	Answer and disconnect supervision are required.

LD 16 – If the application is using the End-of-Dialing timer to transfer outbound calls, the timer must be configured in the Route Data Block.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block
...		
CNTL	YES	Change controls or timers.
- TIMR	EOD 128-(13952)-32640	End-of-Dialing timer in milliseconds. The default is 13952 milliseconds.

LD 17 – In order to originate calls from phantom TNS/DNs, a phantom loop must first be configured and a physical loop card must be installed. A phantom DN can then be configured as part of a specific device group. After configuration changes to the loop card, the system must be reinitialized for the changes to take effect.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CEQU	Release 19 gate opener.
...		
- TERM	0-159 [X] 0-159 [C] 0-159	Single density local terminal loops. Precede loop number with X to remove. Precede loop number with C to create a phantom loop.
- TERD	0-159 [X] 0-159 [C] 0-159	Double density local terminal loops. Precede loop number with X to remove. Precede loop number with C to create a phantom loop.
- TERQ	0-159 [X] 0-159 [C] 0-159	Quad density local terminal loops. Precede loop number with X to remove. Precede loop number with C to create a phantom loop.

LD 97 – If a superloop is used, the phantom loop is configured in this overlay.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	SUPL	Superloop parameters.
SUPL	0-156 [X] 0-156 [C] 0-156	Superloop number in multiples of four. Precede superloop number with X to remove. Precede superloop number with C to create a phantom superloop.

LD 11 – After configuring the phantom loop, an AST Meridian 1 proprietary set can be designated to a specific device group which can be controlled by applications. Therefore, when an application wants to originate a call on behalf of an idle TN, it can use a phantom TN. This idle TN is an AST Meridian 1 proprietary set which is defined on a phantom loop. There is no upper limit on the number of devices per group defined by the Phantom DN. However, there is an upper limit on the number of TNs that can be defined for the loop card. This number is dependent on the density of the loop card. The ITNA and DGRP prompts must be configured as follows:

Prompt	Response	Description
REQ	NEW	New.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. Terminal Number for the Option 11.
...		
CDEN	 SD DD 4D	Card density. Single density. Double density. Quad density.
DES	phanDN	One-to-six character Office Data Administration System (ODAS) Station Designator.
CUST	0-99 0-31	Customer number. For Option 11C.
...		
CLS	NDD	No digit display is recommended if configuring phantom devices.
CLS	(DNDD)	Dialed Name Display denied is recommended if configuring phantom devices.
...		
AST	00	Key 0 is AST.

IAPG	(0)-15	Meridian Link Unsolicited Status Message (USM) group. These groups determine which status messages are sent for an AST set. The default 0 sends no messages, whereas Group 1 sends all messages.
ITNA	(NO) YES	Idle TN for Third Party Application. Set ITNA to YES for Phantom TN calls.
DGRP	(1)-5	Device Group with which phantom TNs are associated.
...		
KEY	xx SCR yyyy	Key number, Single Call Ringing, DN.
CLS	xx RLS	Key number, Release.

Feature operation

Applications invoke the Fast Transfer feature by sending a Fast Transfer request message to the switch. No specific operating instructions are required to use this feature.

Pretranslation

In a business or hospitality environment, many communications situations can be simplified with a flexible dialing plan. Pretranslation lets you create such a plan by using Speed Call lists as Pretranslation Tables.

Some typical uses of Pretranslation are:

— room number to DN correlation

— partitioning of telephones by category, group, department, floor, building, room, or special service

— internal call restrictions, and

— expanded customer dialing capability.

The dialing capabilities and/or restrictions of each Pretranslation group are defined in Pretranslation Tables. The tables are Speed Call lists modified for Pretranslation.

With Pretranslation, only the first dialed digit of a call is pretranslated. The translation choices are:

— **Pass** the digit as dialed with no changes

— **Replace** the first dialed digit with a specified substitute digit or digits, and pass the remaining digits unchanged

— **Delete** the first dialed digit and pass the remaining digits unchanged, or

— **Block** the call based on the first digit dialed.

The pretranslator must deal with all telephones, trunks, and consoles capable of delivering a dialed digit to the Meridian 1 digit processor. Each of these must be assigned to one of 255 Pretranslation groups. The groups are generally set up as follows:

— trunk and Direct Inward System Access (DISA) calls default to group 0

— Attendant Consoles default to group 1, and

— telephones and terminals default to group 0, but may be assigned to groups 2-254.

Note: Before X11 Release 14, there are a maximum of eight Pretranslation groups (0-7).

The dialing capabilities of each group are reflected by the codes stored against entries in the Pretranslation Table. The four possible codes are:

Code	Function
*	Block call.
***	Delete Pretranslation (first dialed) digit, pass remaining digits unchanged.
space <CR>	Pass Pretranslation digit unchanged.
xxxx...x	Pretranslate digit into xxxx...x, where: xxxx...x = replacement DN.

Only the first dialed digit is sent from the digit processor to the pretranslator. The pretranslator looks up the stored code for the dialed digit in the Pretranslation Table associated with the calling terminal, applies the treatment specified by the entry, and passes the result to the DN translator. From then on, the call is processed normally. Pretranslation of the call is finished at this point, unless call modification procedures, such as a Call Transfer, are involved.

Setting up dialing plans and Pretranslation Tables

Steps needed to set up Pretranslation:

1. Identify the customer numbering plan.
2. Determine access and restrictions for each Pretranslation calling group.
3. Determine dialing requirements and instructions for the Pretranslation calling groups and create a Pretranslation Table for each group.
4. Implement the feature.

A hotel has been chosen as a model to illustrate the principles of Pretranslation and how to set up Pretranslation. However, Pretranslation can be applied to many other business environments.

Table 118
Description of model

Hotel with 12 floors containing administrative offices, hotel services, and guest rooms.
Floor 1 – Lobby, gift shop, restaurants, and administrative offices.
Floor 2 – Meeting rooms, salon, and additional office space.
Floor 3 – Banquet rooms and health club.
Floors 4-12 – Guest rooms (floors 4-9 each have 50 rooms, floors 10-12 each have 25 suites).

Step 1 – Identify the numbering plan

The model hotel's numbering plan is shown in Table 119.

Table 119
Numbering plan for model

Available numbers	Assigned to	Actual DNs used
0	Operator	0
1	Guest rooms on floor 10 Guest rooms on floor 11 Guest rooms on floor 12	1001-1026 1101-1126 1201-1226
2	Room service Cafe Restaurant Gift shop Health club Salon Housekeeping Bell Captain Valet Meeting rooms Administrative offices Security Front desk Lobby telephones Miscellaneous	2001 2002 2003 2004 2005 2006 2007 2008 2009 2100-2199 2300-2599 2700 2730 2750-2765 2800-2899
3	SPRE code	
4	unused	
5	unused	
6	Trunk access codes	620-635
7	Guest rooms on floor 4 Guest rooms on floor 5 Guest rooms on floor 6 Guest rooms on floor 7 Guest rooms on floor 8 Guest rooms on floor 9	7401-7451 7501-7551 7601-7651 7701-7751 7801-7851 7901-7951
8	unused	
9	BARS access codes	9

Step 2 – Determine access restrictions

Pretranslation calling groups and dialing restrictions are shown in Table 120.

Table 120
Access and restrictions for model

Group number (XLST)	Type of station	Allowed access	Denied access
0	Default for DISA trunks and telephones	Operator only	All except Operator
1	Guest rooms	Other guest rooms, hotel services, local and long distance, operator	Administrative telephones and direct trunk access
2	Lobby and courtesy telephones	Guest rooms, security, and the operator	Hotel services, administrative telephones, local and long distance, direct trunk access, and SPRE
3	Administrative A	Guest rooms, administrative telephones, direct trunk access, SPRE, operator, BARS access for local and long distance	Direct trunk access
4	Administrative B	Guest rooms, administrative telephones, SPRE, operator	Direct trunk access, BARS access for local and long distance

Step 3 – Determine dialing requirements and create Pretranslation Tables

Dialing instructions for Group 0 (zero) in this model are shown in Table 121 and the corresponding Pretranslation Table is listed in Table 122. For an explanation of the groups used in this model, see Table 120.

Table 121
Group 0 – Default for unassigned trunks and telephones

Actual digits dialed	Desired destination
1	Operator
2	Operator
3	Operator
4	Operator
5	Operator
6	Operator
7	Operator
8	Operator
9	Operator
0	Operator

Table 122
Group 0 – Pretranslation Table (default)

Digit	Code	Function	Destination
1	0	replace	Operator
2	0	replace	Operator
3	0	replace	Operator
4	0	replace	Operator
5	0	replace	Operator
6	0	replace	Operator
7	0	replace	Operator
8	0	replace	Operator
9	0	replace	Operator
0	space <CR>	pass	Operator

Dialing instructions for Group 1 in this model are shown in Table 123 and the corresponding Pretranslation Table is listed in Table 124.

Table 123
Group 1 – Guest dialing instructions for model

Actual digits dialed	Desired destination
1xxx	Guest rooms on floors 10-12
2	Security
3	SPRE (housekeeping staff for Room Status)
4	Front desk
51	Room Service
52	Cafe
53	Restaurant
54	Gift shop
55	Health club
56	Salon
57	Housekeeping
58	Bell captain
59	Valet
7xxx	Guest rooms on floors 4-9
8	Long distance calls
9	Local calls
0	Operator

Table 124
Group 1 – Pretranslation Table (Guests)

Digit	Code	Function	Destination
1	space <CR>	pass	Guest rooms
2	2700	replace	Security
3	space <CR>	pass	SPRE
4	2730	replace	Front desk
5 (see Note)	200	replace	Guest services
6	*	block call	Not used
7	space <CR>	pass	Guest rooms
8	620	replace	Long distance calls
9	space <CR>	pass	Local calls
0	space <CR>	pass	Operator
Note: When a guest dials 51 for room service, the digit "5" is translated to the entry "200" and the 1 is passed as is, resulting in the extension "2001."			

Dialing instructions for Group 2 in this model are shown in Table 125 and the corresponding Pretranslation Table is listed in Table 126.

For an explanation of the groups used in this model, see Table 120.

Table 125
Group 2 – Lobby and courtesy telephone dialing instructions

Actual digits dialed	Desired destination
1xxx	Guest rooms on floors 10-12
2	Security
7xxx	Guest rooms on floors 4-9
0	Operator

Table 126
Group 2 – Pretranslation Table (lobby and courtesy telephones)

Digit	Code	Function	Destination
1	space <CR>	pass	Guest rooms
2	2700	replace	Security
3	*	block call	Not used
4	*	block call	Not used
5	*	block call	Not used
6	*	block call	Not used
7	space <CR>	pass	Guest rooms
8	*	block call	Not used
9	*	block call	Not used
0	space <CR>	pass	Operator

Dialing instructions for Group 3 in this model are shown in Table 127 and the corresponding Pretranslation Table is listed in Table 128.

For an explanation of the groups used in this model, see Table 120.

Table 127
Group 3 – Administrative A dialing instructions for model

Actual digits dialed	Desired destination
1xxx	Guest rooms on floors 10-12
2xxx	Administrative telephones
3	SPRE
7xxx	Guest rooms on floors 4-9
9	Local/long distance through BARS
0	Operator

Table 128
Group 3 – Pretranslation Table (Administrative A)

Digit	Code	Function	Destination
1	space <CR>	pass	Guest rooms
2	space <CR>	pass	Administrative telephones
3	space <CR>	pass	SPRE
4	*	block call	Not used
5	*	block call	Not used
6	*	block call	Not used
7	space <CR>	pass	Guest rooms
8	*	block call	Not used
9	space <CR>	pass	Local/long distance through BARS
0	space <CR>	pass	Operator

Dialing instructions for Group 4 in this model are shown in Table 129 and the corresponding Pretranslation Table is listed in Table 130.

For an explanation of the groups used in this model, see Table 120.

Table 129
Group 4 – Administrative B dialing instructions for model

Actual digits dialed	Desired destination
1xxx	Guest rooms on floors 10-12
2xxx	Administrative telephones
3	SPRE
7xxx	Guest rooms on floors 4-9
0	Operator

Table 130
Group 4 – Pretranslation Table (Administrative B)

Digit	Code	Function	Destination
1	space <CR>	pass	Guest rooms
2	space <CR>	pass	Administrative telephones
3	space <CR>	pass	SPRE
4	*	block call	Not used
5	*	block call	Not used
6	*	block call	Not used
7	space <CR>	pass	Guest rooms
8	*	block call	Not used
9	*	block call	Not used
0	space <CR>	pass	Operator

Operating parameters

Pretranslation Table codes are limited to the four described on page 2320.

User groups are limited to 255 (eight before X11 Release 14).

Each Pretranslation Table entry can be up to 31 characters long; however, it is recommended that a maximum of eight characters be used.

After Pretranslation, any previously loaded (but not pretranslated) digits are added to the end of the pretranslated digits. If the total number of digits exceeds 31, the excess digits will be truncated.

Each Pretranslation Table reduces the number of available Speed Call lists in the system.

Speed Call Controllers do not have access to Pretranslation Tables. Lists must be created and maintained through Service Change.

Before configuring a Pretranslation Data Block in LD 18, Pretranslation group 0 must be configured.

When Pretranslation is allowed in LD 15 (PREO = 1), in order for a pretranslation entry to be removed, Pretranslation must first be disabled in LD 15 (PREO = 0). The Pretranslation data block is then removed in LD 18. It is not possible to remove a single entry. The entire data block must be removed.

Feature interactions

Authorization Code Security Enhancement

The first digit dialed after a valid Authorization Code is sent to the pretranslator.

Automatic Trunk Maintenance
Private Line
Telset Messaging

Pretranslation cannot be used with these features.

Automatic Wake Up

When the Pretranslation feature is equipped with AWU, the actual DN, not the pretranslation DN, should be used when programming the AWU call request.

Call Detail Recording

If a number dialed is pretranslated, the translated digits appear in the Call Detail Recording (CDR) records, not the dialed digits.

Call Forward

The DN dialed-forwarded calls are pretranslated.

Charge Account, Forced

The first digit dialed after a valid Charge Account Code is sent to the pretranslator.

Controlled Class of Service, Enhanced

The DN used to program the Controlled Class of Service (CCOS) should be the actual DN before pretranslation. When programming CCOS, the DN entered is not pretranslated.

Digit Display

The Pretranslation digit is displayed as it was dialed, but if the call is put on hold, the digits of the pretranslated DN are displayed.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

The Pretranslation feature is supported in a DPNSS1 UDP network. At the originating node, the first digit dialed of a call is pretranslated to trigger the look-up of the stored code for the dialed digit in the pretranslation table associated with the calling terminal.

Direct Inward System Access

Direct Inward System Access calls are automatically assigned XLST 0.

Direct Private Network Access

Digits automatically inserted by Direct Private Network Access Digit Insertion are pretranslated during call processing in the same manner as if the caller had manually dialed the digits.

Electronic Switched Network

The pretranslator is used with calls to HNPA, HLOC, and Home CDP locations.

Flexible Feature Codes

Flexible Feature Codes must be accessible through a Pretranslation Table entry in order for users to activate features in this manner.

The Flexible Feature Code (FFC) feature will not be affected if the FFC's begin with "*" or "#", since before translation begins if the first digit is an "*" or "#" pretranslation will not be done. If any digits follow the FFC code, the first of the digits that follows will be pretranslated.

Meridian Hospitality Voice Services

Prior to Meridian Hospitality Voice Services (MHVS), the setup of calls using the Applications Module Link (AML) was not supported from telephones using the Pretranslation feature. With MHVS equipped, call setup using the AML is supported.

Meridian Link Calls

Pretranslation cannot function with Meridian Link calls if the Hospitality Voice Services (HVS) package is enabled.

Special Prefix

The SPRE code must be accessible through a Pretranslation Table entry in order for users to activate features in this manner.

Speed Call
Speed Call, System

With X11 Release 18 and later, a Speed Call List number should be programmed to allow for Pretranslation. For example, if 9 pretranslates to 99 and you want to reach 99 nxx xxxx, you need to program the number in the Speed Call List as 9 nxx xxxx. When the Speed Call List is used, 9 nxx xxxx is pretranslated at call processing time to become 99 nxx xxxx.

User Selectable Call Redirection

If Pretranslation (package 92) is enabled, the digits entered as the redirection DN are pretranslated before they are stored. Note that no Pretranslation occurs when the redirection DNs are used in such call processing features as Hunting or CFNA, eliminating the possibility that the redirection DN is pretranslated twice.

Feature packaging

Pretranslation (PXLT) package 92 has no feature package dependencies.

Feature implementation

LD 17 – Allocate sufficient Speed Call lists to be used as Pretranslation Tables (X11 Release 13 and later software).

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PARM	Configuration Record. Release 19 gate opener.
...		
- MSCL	(0)-8191	Maximum number of Speed Call lists.

LD 18 – Add or change a Speed Call list to be used for each Pretranslation calling group.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	SCL	Speed Call data block.
LSNO	0-8190	Number of Pretranslation list. ***Note:*** With X11 Release 12 and earlier, up to 255 Pretranslation lists are allowed.
DNSZ	4-(16)-31	Number of digits that can be in a list entry.
SIZE	10	Maximum number of entries.
WRT	(YES), NO	Data is correct and can be updated in data store.
STOR		x is the first digit dialed.
	x *	* = block call.
	x ***	*** = delete the digit.
	x space <CR>	space <CR> = pass digit unchanged.
	x yyyy...y	yyyy...y = replacement digits.
WRT	(YES), NO	Data is correct and can be updated in data store.
STOR	<CR>	Ends input of list entries.

LD 18 – Add or change the Pretranslation data block, defining the calling group to Speed Call list correlation. This procedure is necessary in X11 Release 14 and later software. This list must be configured before Pretranslation (PREO) is enabled in LD 15.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	PRE	Pretranslation (X11 Release 14 and later software).
CUST	0-99 0-31	Customer number. For Option 11C.
XLAT	xxx yyyy	Pretranslation list, where: xxx = Pretranslation calling group number (0-254), and yyyy = corresponding Speed Call list number (1-8190). ***Note:*** XLAT appears in LD 15 in X11 Release 13 and earlier software.

LD 15 – Activate Pretranslation and define calling groups to Speed Call list correlation.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
CUST	0-99 0-31	Customer number. For Option 11C.
- PREO	0 1	Allow or deny Pretranslation, where: 0 = no Pretranslation, and 1 = Pretranslation.

Note: When Pretranslation group 0 is configured, care must be taken to define the XLST prompt, rather than letting it default automatically to 0. If XLST does default to 0 when Pretranslation group 0 is configured, all sets in the switch are affected.

LD 10 – Associate a analog (500/2500 type) telephone with a Pretranslation group.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
XLST	0-254	Associate telephone with the specified Pretranslation group (0-7 in X11 Release 14 and earlier).
	<CR>	Default to Pretranslation group 0 (only when REQ = NEW).

LD 11 – Associate a Meridian 1 proprietary telephone with a Pretranslation group.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	aaaa	Telephone type, where:
		aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
XLST	0-254	Associate telephone with the specified Pretranslation group (0-7 in X11 Release 14 and earlier).
	<CR>	Default to Pretranslation group 0 (only when REQ = NEW).

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	23.0x
Networking:	No

Pretranslation and System Speed Call Enhancement

Pretranslation and System Speed Call Enhancement provides the option to allow or deny Pretranslation when a System Speed Call list entry is dial accessed.

The existing Pretranslation feature allows the creation of a flexible dialing plan by using Speed Call lists which are modified for pretranslation. The dialing capabilities and/or restrictions of each Pretranslation group are defined in Pretranslation Tables.

The existing System Speed Call feature allows abbreviated dialing and also allows users to temporarily override the set's Class of Service, Trunk Group Access Restrictions (TGARs), and Code Restrictions.

Analog (500/2500 type) sets, Meridian 1 proprietary sets, and Attendant Consoles can activate System Speed Call by using a Special Prefix (SPRE) or Flexible Feature Code (FFC).

For further information pertaining to the existing Pretranslation and System Speed Call features, refer to the feature modules in this guide.

Tables 131 and 132 are examples of a Pretranslation Table and a System Speed Call list respectively.

Table 131
Pretranslation Table

List entry	Corresponding DN or Code	Function
0	space <CR>	Pass Pretranslation digit unchanged
1	space <CR>	Pass Pretranslation digit unchanged
2	space <CR>	Pass Pretranslation digit unchanged
3	space <CR>	Pass Pretranslation digit unchanged
4	space <CR>	Pass Pretranslation digit unchanged
5	space <CR>	Pass Pretranslation digit unchanged
6	space <CR>	Pass Pretranslation digit unchanged
7	8000	Convert to Route Access Code 8000
8	***	Delete Pretranslation (first dialed) digit, pass remaining digits unchanged
9	*	Block the call

Table 132
System Speed Call List

List entry	Corresponding DN
00	7182
01	122455678
...	...

In Table 131, if the first dialed digit is 0 to 6, Pretranslation passes all of the digits and leaves them unchanged. If the first dialed digit is 7, Pretranslation changes digit 7 to Route Access Code 8000. If the first dialed digit is 8, Pretranslation deletes the first dialed digit and passes the remaining digits unchanged. If the first dialed digit is 9, Pretranslation blocks the call.

To dial access System Speed Call lists, the user dials:

1 SPRE, as defined in Overlay 15

2 System Speed Call Feature Code - 73

3 System Speed Call list entry number

If the Meridian 1 is equipped with Flexible Feature Codes, the user dials:

1 FFC, as defined in Overlay 57.

2 System Speed Call list entry number

With the existing Pretranslation and System Speed Call features, when Dial Access occurs, Pretranslation is performed on the first dialed digit of the Special Prefix (SPRE) or Flexible Feature Code (FFC). The first digit of the digits stored in the System Speed Call list entry is then also pretranslated.

The Pretranslation and System Speed Call Enhancement introduces the BPSS prompt in Overlay 15. This prompt provides the option to allow or deny pretranslation on the System Speed Call list entry when dial accessed. If BPSS is set to YES in Overlay 15, Pretranslation is blocked. Therefore, only the first dialed digit is pretranslated. The first digit of the digits stored in the System Speed Call list entry is not pretranslated.

To follow are examples of Pretranslation and System Speed Call functionalities when Pretranslation is blocked/not blocked. Tables 131 and 132 are considered for these examples. It is assumed that the SPRE method of dialing is used and that the user has the following configuration:

— Special Prefix (SPRE) code - **1**

— System Speed Call Feature Code - **73**

BPSS = NO

With dial access and the BPSS option set to NO in Overlay 15, Pretranslation is not blocked. Therefore, the existing Pretranslation functionality is retained.

Now when the user dials 1+73+00, Pretranslation occurs twice. It occurs once on the first dialed digit (1) and once again on the first digit of the digits stored in the System Speed Call list entry (7 of 7182).

When the user dials SPRE + 73 + 00, the first digit of the digits stored in the System Speed Call list entry (7 of 7182) is converted to DN 8000. In this example, DN 8000 is a Trunk Route Access Code; therefore, the call goes out on that route, and the digits 182 are outpulsed.

BPSS = YES

With dial access and the BPSS option set to YES in Overlay 15, Pretranslation is blocked. Therefore, the new Pretranslation functionality is in effect.

Now when the user dials 1+73+00, the first dialed digit (1) is pretranslated. However, the first digit of the digits stored in the System Speed Call list entry (7 of 7182) is not pretranslated.

When the user dials SPRE + 73 + 00, the list entry number is converted to DN 7182. When BPSS = YES, Pretranslation is blocked at this point. Therefore, the first digit of the digits stored in the System Speed Call list entry (7 of 7182) is not converted to the corresponding DN (8000) in the Pretranslation Table.

Operating parameters

To allow or deny Pretranslation on a System Speed Call list entry when dial accessed, the BPSS prompt must be defined in the Customer Data Block.

When Pretranslation is disabled (PREO = 0) in the Customer Data Block, BPSS is prompted but does not take effect. Therefore, the current functionality is retained.

With Dial Access and the BPSS option set to YES in the Customer Data Block, only the first dialed digit is pretranslated. The first digit of the digits stored in the System Speed Call list entry are not pretranslated.

With Dial Access and the BPSS option set to NO in the Customer Data Block, the existing operation is retained.

The functionality of the Speed Call (Dial Access and Key Access) and System Speed Call (Key Access only) features is not changed by this enhancement.

The operation of Key Access to System Speed Call with Pretranslation is not modified with this feature.

Existing dialing plans are affected when the Pretranslation and System Speed Call Enhancement is configured.

The pre-programmed DN in the System Speed Call list can be internal or external to the Meridian 1 system.

Feature interactions

There are no new feature interactions as a result of this enhancement. For a list of the existing Pretranslation, Speed Call, and System Speed Call feature interactions, refer to the Pretranslation, Speed Call, and System Speed Call feature modules in *X11 features and services*.

Feature packaging

The following packages are required for Pretranslation and System Speed Call Enhancement:

— System Speed Call (SSC) package 34

— Pretranslation (PXLT) package 92

Feature implementation

Note: The Pretranslation and System Speed Call features must be configured as per the existing implementation procedures. Refer to the Pretranslation feature module and the System Speed Call feature module in this guide.

CAUTION

Care must be taken when implementing Pretranslation and System Speed Call Enhancement, as existing dialing plans will be impacted when BPSS = YES. In this case, the existing Pretranslation functionality is changed, and the entire Customer group of dial access System Speed Call users is affected.

LD 15 – Allow or deny blocking of Pretranslation on list entry when dial accessed.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options data. Release 21 gate opener.
CUST	xx	Customer number.
...		

PREO	1	Pretranslation Option enabled. 0 = Pretranslation Option disabled (default).
BPSS	YES	Block Pretranslation on System Speed Call lists when dial accessed. NO = Do not block Pretranslation on System Speed Call lists when dial accessed (default).

Feature operation

To dial access System Speed Call lists, the user

1 lifts the handset of the analog (500/2500 type) set, Meridian 1 proprietary set, or Attendant Console

2 dials the Special Prefix (SPRE) code, as defined in Overlay 15

3 dials the System Speed Call Feature Code - **73**

4 dials the System Speed Call list entry number

If the Meridian 1 is equipped with Flexible Feature Codes (FFCs), the user

1 lifts the handset of the (500/2500 type) set, Meridian 1 proprietary set, or Attendant Console

2 dials the Flexible Feature Code (FFC) for accessing System Speed Call, as defined in Overlay 57

3 dials the System Speed Call list entry number

Introduced in X11 Release:	18.20H
Networking:	Yes

Preventing Reciprocal Call Forward

This feature provides a modification to the Call Forward All Calls feature as a customer option. If set A attempts to enter a new Call Forward All Calls to set B, this modification verifies that set B has not been call forwarded to set A.

The verification process is repeated until one of the following conditions is met:

— the entered DN is not call-forwarded to any other set

— the activating set call forwards to the original Call Forward DN

— the maximum number of hunt steps is encountered (32 steps for NT and XT machines, and 16 steps for ST machines)

— a trunk is encountered, or

— a Pilot DN is encountered.

If a Multiple Appearance DN is encountered during the verification process, the only possible Call Forward Chain is checked.

Operating parameters

Although introduced in X11 Release 16, the ability to allow or deny Preventing Reciprocal Call Forward for a customer by programming LD 15 was introduced in X11 Release 18.20H.

The verification is done only to current Call Forward states of the DNs being checked.

A set cannot Call Forward to itself.

This modification does not apply:

— to Hunt DNs

— to calls forwarded to the attendant, or

— across trunks.

This feature applies to network environments.

Feature interactions

Network Call Redirection

For Network Call Redirection, when a call forwarding loop from one node to another occurs, the maximum number of redirections can be defined by the customer.

Remote Call Forward

This modification applies to Remote Call Forward.

Feature packaging

Preventing Reciprocal Call Forward is included in base X11 system software.

Feature implementation

LD 15 – Allow or deny Preventing Reciprocal Call Forward for a customer:

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
...		
OPT	(PVCA) PVCD	Enter PVCD to (allow) deny Preventing Reciprocal Call Forward.

Feature operation

If set A attempts to enter a new Call Forward All Calls to set B, verification is given that set B has not been call forwarded to set A.

When this situation is encountered:

— If the attempt to enter the new Call Forward DN was made on set A using a SPRE or Flexible Feature Code (typically on a 500/2500-type set), overflow tone is given to set A and the existing call-forward DN remains unchanged.

— If the attempt to enter the new Call Forward DN was made on set A using the Call Forward All Calls feature key, the attempted entry is treated like a normal invalid DN entry (i.e., when the Call Forward All Calls key is pressed a second time after the DN has been entered, the associated lamp continues to flash until a valid forward DN is entered or the key is pressed for a third time).

Privacy

Meridian 1 proprietary telephones automatically provide Privacy for telephones sharing a single call arrangement Directory Number (DN). When a call is in progress on the DN, no other telephone on which the DN appears can enter the call.

Operating parameters

Privacy is not available for analog (500/2500 type) telephones.

If the Directory Number (DN) is shared with any single line telephone, Privacy is not in effect for any appearance of the DN, and anyone sharing that DN can enter an active call.

Feature interactions

Bridging

Privacy is lost when telephones are bridged. Any appearance of the DN can enter the call by going off-hook.

Call Hold, Permanent

A call placed on Permanent Hold has Privacy removed. Privacy is reinstated when the call is removed from Permanent Hold.

Multiple Appearance Directory Number

If a Multiple Appearance, Single Call Arrangement (SCR) or Single Call Arrangement without Ringing (SCN) DN is shared by Meridian 1 proprietary telephones only, Privacy is in effect. No one can enter a call unless the call is first placed on Hold, or unless Privacy Release is activated to allow another appearance to enter the call. If this configuration is shared between these telephones and single-line telephones, Privacy is not in effect for any appearance of the DN. Anyone sharing the DN can enter the call at any time.

Privacy Override

The user can Override the inherent privacy on Meridian 1 proprietary telephones. If an appearance occurs on a telephone with Privacy Override enabled, that appearance can bridge into an active call. This pertains to calls on a multiple appearance single call Directory Number (DN) when not mixed with single line telephones.

Feature packaging

Privacy is included in base X11 system software.

Feature implementation

No change to existing configuration is required for the Privacy feature.

Feature operation

No specific operating procedures are required to use this feature.

Privacy Override

A Meridian 1 proprietary telephone with a Privacy Override Allowed (POA) Class of Service can enter an established call on a multiple appearance single call Directory Number (DN). However, the call cannot be joined until it is established (that is, the EOD timer has expired).

If all members of a non-mixed multiple appearance single call DN group are allowed Privacy Override, the operation of the feature is equivalent to a mixed multiple appearance single call arrangement.

When a group contains a combination of Privacy Override Allowed (POA) and Privacy Override Denied (POD) Classes of Service, the telephones denied Privacy Override cannot bridge into established calls.

Operating parameters

Privacy Override does not apply to analog (500/2500 type) telephones.

Feature interactions

Call Park
Call Transfer

Calls in a Privacy Override conference state cannot be parked or transferred.

Conference

The Conference feature can be used to add other parties to a Privacy Override connection.

Exclusive Hold

Telephones with POA Class of Service cannot bridge into calls on Directory Numbers (DNs) with Exclusive Hold active.

Multiple Appearance Directory Number - Mixed Mode

Since the Privacy feature is not active in this mode, telephones with a POD Class of Service can bridge into an active call.

Privacy

The user can Override the inherent privacy on Meridian 1 proprietary telephones. If an appearance occurs on a telephone with Privacy Override enabled, that appearance can bridge into an active call. This pertains to calls on a multiple appearance single call Directory Number (DN) when not mixed with single line telephones.

Feature packaging

Privacy Override is included in base X11 system software.

Feature implementation

LD 11 – Allow or deny Privacy Override on a Meridian 1 proprietary telephone.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	POA (POD)	Allow or deny Privacy Override.

Feature operation

To activate Privacy Override, press the multiple appearance single call DN. You are automatically connected to the call.

Introduced in X11 Release:	All
Networking:	No

Privacy Release

In multiple appearance single call arrangements of Meridian 1 proprietary telephones, Privacy Release allows one other appearance of the Directory Number (DN) to enter the call. Privacy is then reestablished until Privacy Release is activated again.

Operating parameters

Privacy Release is available only with Meridian 1 proprietary telephones in multiple appearance single call arrangements.

The system must be equipped with a conference loop.

Feature interactions

Automatic Redial

When an Automatic Redial (ARDL) call is not accepted by the calling party, the Privacy Release (PRS) key is ignored.

Call Park

When a call from a Meridian 1 proprietary telephone is parked, that telephone cannot activate Privacy Release. For example, Party A calls Party B. Party B parks the call. Party A cannot activate Privacy Release.

China – Attendant Monitor

If Privacy Release is activated on a set that is involved in a monitored call, Attendant Monitor is deactivated.

Dial Access to Group Calls
Group Call

The Privacy Release feature cannot be applied to Dial Access to Group Calls and Group Call.

Exclusive Hold

If the telephone with Privacy Release has Exclusive Hold Allowed in the Class of Service, and a call is on hold, another telephone with that Multiple Appearance Directory Number (MADN) cannot access the call.

Multiple Appearance Directory Number

Privacy Release has no effect on Multiple Appearance, Multiple Call Arrangement with Ringing (MCR), or Multiple Call Arrangement without Ringing (MCN) calls.

Music, Enhanced

When using Privacy Release to add one or more members to a call already receiving Music, the Music is removed.

Ring and Hold Lamp Status

If the Privacy Release feature is activated for multiple-appearance single-call DNs, the blinking rate is based on the Class of Service of each set on which other appearances of the DN occur.

Feature packaging

Privacy Release is included in base X11 system software.

Feature implementation

LD 11 – Allow/deny Privacy Release for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx PRS	Add a Privacy Release key. M2317 and M3000 telephones automatically assign the PRS key to key 28.

Feature operation

To allow someone with another appearance of the Directory Number (DN) to enter a call:

1 Press **Priv Rls**. All appearances of that DN flash. One other party can enter the call by pressing the flashing DN key that has the call.

2 You must press **Priv Rls** again to allow another appearance of the DN to enter the call.

Private Line Service

Private Line Service enables the customer to assign private Central Office (CO) lines to selected telephones or power fail transfer equipment. When associated with a Meridian 1 proprietary telephone, the following features are available to Private Line Service:

- Automatic Dialing
- Automatic Preselection
- Call Pickup
- Call Transfer
- Call Status
- Conference
- Common Audible Signaling
- Hold
- Multiple appearance single call arrangement
- Prime Directory Number
- Privacy
- Privacy Release
- Release, and
- Analog (500/2500 type) telephone/SL-1 telephone mix.

Operating parameters

Single line telephones with Private Line Service cannot access Meridian SL-1 features.

All Private Lines must be assigned to trunk route 31 on X11 Release 13 and earlier software. A Directory Number (DN) must be assigned to each trunk.

A maximum of 126 Private Lines are available per customer.

X11 Release 14 and later software allows 512 Private Line trunk routes to be defined.

A Private Line should not be assigned as a Prime Directory Number (DN) unless preselection is required.

Hunting does not apply to Private Line service.

Call Forward on Private Lines (Meridian 1 proprietary telephones) is not forwarded to a second appearance of its own DN.

Feature interactions

Call Modification Features (CMF) in the trunk data block can be inhibited as follows:

— Call Transfer

— Conference

— Call Forward, and

— Message Center.

— Call Forward No Answer
Call Forward No Answer is always inhibited on Private Lines.

— Multiple appearance
For multiple appearance calls, call modification cannot be blocked.

Automatic Line Selection

A Private line DN is selected by Incoming Ringing/Non-Ringing Line Selection and Outgoing Line Selection.

Automatic Redial

An Automatic Redial (ARDL) call can be activated on a Private Line Service key. The call can only be redialed when the calling party's PVR or PVN key is free.

Call Park

Private lines cannot park a call.

Calling Party Privacy

The Private Line Service feature will outpulse the Privacy Indicator only if it is dialed by the originator. An asterisk will be outpulsed to the far end only if it is an Outpulsing of Asterisk and Octothorpe (OPAO) call, otherwise the asterisk signals a three-second pause.

China – Attendant Monitor

Attendant Monitor is blocked from monitoring a Private DN.

Collect Call Blocking

If an incoming DID or CO call from a private line trunk terminates on a set with a CCBA Class of Service, the Collect Call Blocking answer signal is provided in place of the regular answer signal.

Do Not Disturb

Do Not Disturb cannot be used on Private Lines.

Flexible Feature Code Boss Secretarial Filtering

Flexible Feature Code Boss Secretarial Filtering takes precedence over Private Line and Hot Line.

Hot Line

A Hot Line key cannot be a Private Line, as this would defeat the benefits of Private Line service.

Station-to-Station Calling

You must go over the public network to reach a Private Line. The software PRDN is not meant to be dialed directly.

Feature packaging

Private Line Service is included in base X11 system software.

Feature implementation

LD 16 – Add or change a Private Line trunk route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	0-99 0-31	Customer number. For Option 11C.
ROUTE	0-511 0-127	Route number. For Option 11C.
TKTP	COT	Central Office trunk.
AUTO	(NO) YES	Trunks in this route autoterminate.
ICOG	IAO	Incoming and outgoing route.

LD 14 – Add or change Private Line trunks in the Private Line trunk route.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	COT	Central Office trunk.
TN	l s c u c u	Terminal Number. For Option 11C.
XTRK	XUT XEM	Universal Trunk Card (NT8D14), E&M Trunk Card (NT8D15). Prompted only for Superloops and the first unit on the card.
PRDN	xxx...x	Private Line phantom DN.
CMF	(NO) YES	Call modification is or is not inhibited for private line.

LD 10 – Add or change Private Line Service for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
DN	xxx...x	Private Line DN (xxx...x is the same as for PRDN prompt in LD 14).

LD 11 – Add or change Private Line Service for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx PVN yyy...y	Private Line non-ringing key (yyy...y is the same as for PRDN prompt in LD 14).
	xx PVR yyy...y	Private Line ringing key (yyy...y is the same as for PRDN prompt in LD 14).

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	10
Networking:	No

Property Management System Interface

The Property Management System Interface (PMSI) is a full-duplex RS-232 asynchronous data link that allows a Meridian 1 customer with a Property Management System (PMS) computer to exchange a higher level of protocol for the Background Terminal (BGD) features in a hospitality environment.

The Meridian 1 sends formatted messages to the Property Management System (PMS) computer for the following features:

— Controlled Class of Service (CCOS)

— Message Waiting

— Do Not Disturb (DND)

— Room Status (RMS)

— Call Number Information Messages (CNIM), and

— Call Party Name Display (CPND).

The system connects to the Property Management System (PMS) computer through a Serial Data Interface (SDI) port. Each character received from the Property Management System Interface (PMSI) data link is treated as if it were entered from a TTY, and each character transmitted to the PMS computer is handled the same way as characters output to a TTY.

PMSI Standardization

The PMSI Standardization features in X11 Release 19 and later provide the Meridian 1 with the following enhancements:

— message retransmission

— polling, and

— message monitoring.

Note: Upon loading X11 Release 19, these features are not automatically activated. You must go into LD 17 to enable these features.

Message transmission and retransmission

Prior to X11 Release 19, the Meridian 1 ignored any response returned by the PMS after sending a room status message to the PMS, and did not attempt to retransmit the message. As a result, the database between the PMS and the Meridian 1 could not be maintained consistently.

With X11 Release 19 and later, PMSI Standardization provides the Meridian 1 with the capability to retransmit a message to the PMS. This means whenever the Meridian 1 transmits a room message or the new polling message to the PMS, the Meridian 1 will wait for an <ACK> response from the primary PMSI port. If the Meridian 1 receives a <NAK>, or does not receive any response before the predefined response timer expires, the same message will be retransmitted to the primary PMSI port.

Polling

Prior to X11 Release 19, the Meridian 1 did not have the capability to monitor the status of the PMSI link (i.e., the link between the Meridian 1 and the PMS).

With X11 Release 19 and later, PMSI Standardization provides this monitoring capability by enabling the Meridian 1 to poll the PMSI link at predefined intervals.

Message monitoring

Prior to X11 Release 19, the Meridian 1 did not have the capability to track incoming messages from the PMS or outgoing messages to the PMS.

With X11 Release 19 and later, PMSI Standardization provides this tracking capability by enabling these incoming/outgoing messages between the Meridian 1 and the PMS to be displayed on all maintenance (MTC) TTYs on the Meridian 1.

Refer to *Property Management System Interface description* (553-2801-101) for detailed information on PMSI Standardization.

Operating parameters

Refer to *Property Management System Interface description* (553-2801-101).

Feature interactions

Refer to *Property Management System Interface description* (553-2801-101).

Feature packaging

Property Management System Interface (PMSI) package 103 requires:

— Controlled Class of Service (CCOS) package 81

— Room Status (RMS) package 100, and

— Background Terminal Facility (BGD) package 99.

Note: PMSI Standardization requires Release 19 software.

Feature implementation

Refer to *Property Management System Interface description* (553-2801-101).

Feature operation

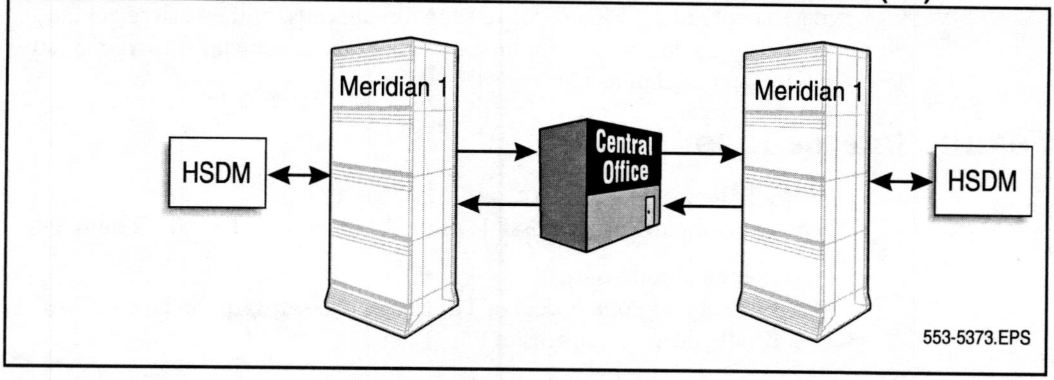
No specific operating procedures are required to use this feature.

Introduced in X11 Release:	19
Networking:	Yes

Public Switched Data Service

The Public Switched Data Service (PSDS) allows you to receive data on your Meridian 1 at 56 kbps over Digital Trunk Interface (DTI) trunks (with X11 Release 16 and later), and at 64 kbps over an Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) channel (with X11 Release 18 and later). See Figure 72.

Figure 72
Public Switched Data Service (PSDS) between Meridian 1 and Central Office (CO)



You can install a T1 link to different vendors and use the Meridian Communications Adapter (MCA) or QMT21 High Speed Data Module to initiate or receive a 56 kbps digital data call. The digital data call then transports across the vendor's digital network to another Meridian 1 or an SL-100.

Operating parameters

Public Switched Data Service (PSDS) requires X11 Release 16 or later. The various data modules are supported for different Releases. The Meridian Communications Adapter (MCA) operates with X11 Release 16 and later. The QMT21 module operates with X11 Releases 16 and 17.

PSDS calls are supported in the following situations:

— a Meridian 1 and the Central Office (CO)

— a tandem call from an SL-100 to a Meridian 1, and

— a Meridian 1 and other PSDS-compatible switches.

The PSDS supports Digital Trunk Interface (DTI) type trunks, TIE and DID/DOD trunks, and Electronic TIE Network (ETN) compatible signaling.

End-to-End DTI network

For all Meridian 1 networks (Point to Point), users can access the existing data facility in the Meridian 1 to support data calls, or they can select the Switched 56 data mode. For mixed-vendor private networks, users can only select the PSDS mode.

Feature interactions

ISDN PRI

The following routes are possible using this feature on Primary Rate Access:

— Point to Point access
  For Point to Point access of TIE trunks, the software can be modified to handle the requirements of this feature.

— Tandem call
  For tandem access, additional information on this feature is needed, or the data call can be defined as a voice call.

— DID/FEX/WATS/Accunet
  The Meridian 1 supports PSDS data calls to these trunk types.

— Public Network hop off
  Signaling is provided to inform the tandem switch about the PSDS data call.

Feature packaging

Public Switched Date Service is included in base X11 system software.

Feature implementation

The data selection (DSEL) in the Route Data Block can be defined as voice calls only (VCE), data calls only (DTA), or voice and data calls (VOD). The call can be defined as voice calls, regular data calls, or PSDS calls. Refer to the *X11 input/output guide* to configure the Route Data Block.

Feature operation

Originating data calls

For direct access, dial the regular seven-digit or 10-digit number.

For special route access, dial a route access code after hearing a dial tone.

Receiving data calls

Calls are answered automatically.

An auto-answer call is answered by the data module, and no special operation is necessary.

Related features

When using PSDS, you may want to refer to the following features.

Meridian Communications Adapter (MCA)

The Meridian Communications Adapter (MCA) operates with X11 Release 16 and later and allows asynchronous ASCII terminals, personal computers, and printers to be connected to the telephone using an RS-232C or V.35 interface. With Release 14 and later, the MCA also allows synchronous applications (DTEs such as video conferencing equipment and Group IV fax units) to be connected to the telephone. Refer to *Meridian Communications Unit and Meridian Communications Adapter description, installation, administration, operation* (553-2731-109) for detailed information on the MCA.

Meridian Communications Unit (MCU)

The Meridian Communications Unit (MCU) is a X11 Release 19 feature that provides a standalone version of the Meridian Communications Adapter (MCA).

The Meridian Communications Unit (MCU) allows you to transmit and receive data using either PSDS over the public network or a private network. The MCU, which replaces the QMT21C, is designed for domestic and international use, with transmission speeds up to 19.2 kbps asynchronous, and 64 kbps synchronous, integrated display, and self diagnostics. The MCU supports autodialing, ring again, and speed calling, as well as autobauding and automatic parity detection. You can use the MCU for:

— Video conferencing

— LAN bridging

— Bulk data/PC file transfer

— Dial back-up, and

— Host connectivity.

The MCU fully complies with RS-232C and can be configured as DCE or DTE to connect to a terminal, printer, or fax machine.

Unlike the MCA, the MCU provides a dedicated call key and call progress tones. The MCU also permits smart modem pooling.

The MCU supports the DM-DM, T-Link, V.25 bis, and PSDS interfaces as well as the RS-232C, CCITT V.35, CCITT V.24, and RS570/RS3449 (with different cables) interfaces. It complies with V.28 for European approval.

Refer to *Meridian Communications Unit and Meridian Communications Adapter description, installation, administration, operation* (553-2731-109) for detailed information on this feature.

Transparent Data Networking (TDN)

Transparent Data Networking is an X11 Release 19 feature that provides a transparent data channel for data modules to perform end-to-end protocol exchange. This means that two data modules will wait for a circuit path to be established before exchanging protocol parameters.

The data modules and protocols that are supported by TDN are:

— Meridian Communications Adapter (MCA) card in a Meridian Modular telephone (MMT) set, which uses PSDS and T-Link protocols on external calls

— Meridian Communications Unit (MCU), a standalone version of the MCA, which uses T-Link and PSDS protocols on external calls

— Basic Rate Interface (BRI) telephones, which use T-Link, V.110, and V.120 protocols, and

— High Speed Data Module (HSDM) when configured to use PSDS.

Refer to *Transparent Data Networking* (553-2731-110) for detailed information on TDN.

Pulsed E&M DTI2 Signaling

This feature provides pulsed channel associated ABCD-bit line signaling on 2 Mbps digital trunks. This signaling is used by the French Colisée and Indonesian systems, and is equivalent to analogue pulsed E&M signaling. Pulsed E&M 2 Mbps Digital Trunk Interface (DTI2) signaling can be configured by using LD 16 and 73.

Operating parameters

This feature does not apply to Option 11 systems.

Firmware changes to the QPC915C (French Colisée Pulsed E&M DTI2 signaling pack) and the QPC536E DTI (Indonesian Pulsed E&M DTI2 signaling pack), to implement the timing requirements of successive signals for both French Colisée and Indonesia.

Feature interactions

China Number 1 signaling

Cancel Offering (Toll Operator Break Out) is added to the Toll Operator Break-in feature. Calling Party Control is enhanced to use the OHTT, as well as the OHT prompt in LD 16.

Digital Trunk Interface (DTI) – Commonwealth of Independent States (CIS)

Pulsed E&M is not supported by CIS DTI.

2 Mbps Digital Trunk Interface

Pulsed E&M DTI2 signaling is based on 2 Mbps DTI.

MFE for Socotel

Pulsed E&M DTI2 signaling is compatible with MFE for Socotel in the slave mode.

MFC/Semi-compelled MFC

Pulsed E&M DTI2 signaling is compatible with MFC and Semi-compelled MFC (SMFC).

New Toll Call Identification

Pulsed E&M DTI2 signaling is used to distinguish between national and international calls, in order to initiate clear back timing of the correct duration.

Periodic Pulse Metering

Pulsed E&M DTI2 signaling provides the following changes to PPM:

— the ANSWER and RE-ANSWER signals will be counted as a PPM pulse

— the counting of PPM pulses will not be activated when the call is set up; it will be activated when an ANSWER or RE-ANSWER signal is received, and

— PPM pulse detection will be turned off when a CLEAR BACK signal is received.

Lockout

Pulsed E&M DTI2 signaling will allow a flexible treatment to occur on outgoing trunks which are locked out. This will consist of allowing outgoing trunks which are locked out to send repeated FORWARD RELEASE signals.

Feature packaging

Pulsed E & M DTI2 Signaling requires the following packages:

— Pulsed E&M (PEDM) package 232

— International Supplementary Features (SUPP) package 131

— 2 Mbps Digital Trunk Interface (DTI2) package 129

— Special Services for 2500 Sets (SS25) package 18

— 500 Set Dial Access to Features (SS5) package 73

— Operator Call Back (China #1) (OPCB) package 126

— Attendant Break-in/Trunk Offer (BKI) package 127

— PPM/Message Registration (MR) package 101

— Multifrequency Compelled Signaling (MFC) package 128

Feature implementation

LD 16 – Configure the Route Data Block for Pulsed E&M DTI2 Signaling.

Prompt	Response	Description
...		
RPPM	...	
A1MR		First Meter Pulse. Prompted if DTRK = YES, DGTP = DTI2 and MR = PPM.
	(NO) YES	Enter YES to cause the meter pulses received before an ANSWER signal to be invalid. The ANSWER signal is taken as the start of the first charging period (i.e., when an ANSWER signal is received, the PPM count is incremented). NO is the default, and causes the meter pulses to be counted from the moment that the outgoing trunk is seized. When the trunk answers, the PPM count is left unchanged.
...		
IMCB	...	
TOBO		Toll Operator Break Out. Prompted if DTRK = YES, DGTP = DTI2 and MR = PPM.
	(NO) YES	If YES is entered, an OPCA signal received after a toll operator Break-in operation has been completed will result in the toll operator being removed off the call. If NO (the default) is entered, OPCA signals after a toll operator Break-in operation will be ignored.
...		
IHT	...	

OHT	0-(30)-62	Prompted if CNTL = YES and OPCB = YES. Enter the number of seconds, in increments of two, after which an outgoing CGPC non-toll call will disconnect, after the far end disconnects.
OHTT	0-(30)-62	Prompted if CNTL = YES and OPCB = YES. Enter the number of seconds, in increments of two, after which an outgoing CGPC toll call will disconnect, after the far end disconnects.
...		
FALT	...	
FRIN		Forward Release Indefinitely. Prompted only if DTRK = YES and DGTP = DTI2.
	(NO) YES	If YES is entered, a FORWARD RELEASE signal is re-sent every time the Disconnect Supervision timer expires and every time it is restarted. If NO (the default) is entered, a FORWARD RELEASE signal is not resent.
FRRC		Forward Release Repetition Count. Prompted only if FRIN = YES.
	0-(4)-15	Enter the value for the number of times that FORWARD RELEASE signal is resent before an error message is printed, if an acknowledgment is expected but not received.
FRRS		Forward Release Repetition Seize. Prompted only if FRIN = YES.
	(NO) YES	Enter YES to reseize the trunk before resending the FORWARD RELEASE signal. Enter NO to not have the trunk reseized before the FORWARD RELEASE signal is resent.

FRRD	128-(384)-1920	Forward Release Repetition Delay, in milliseconds. This is the delay between sending the SEIZE signal and FORWARD RELEASE signal. It is only prompted if FRIN = YES and FRRS = YES.
RRBS		Repeat Release Before Seize. This prompt allows a FORWARD RELEASE signal to be sent immediately before a SEIZE signal on a DTI2 trunk. Prompted only if DTRK = YES, DGTP = DTI2, and FRRS is not set to YES.
	(NO) YES	Enter YES to have a FORWARD RELEASE signal resent followed by the SEIZE signal. Enter NO to seize the trunk normally.
RLSM	(0)-15	Release Mechanism Only prompted if DTRK = YES and DGTP = DTI2.

LD 73 – Configure the DTI Data Block for Pulsed E&M DTI2 Signaling.

Prompt	Response	Description
...		
PERS	...	
DBNC	(10)-32	The De-bounce time for ABCD bit signals.
...		
TIME	...	
MINP	(8)-256	The Minimum Pulse Length for a Meter Pulse.
SASU	0-(1920)-32256	The Seize Acknowledge Supervision time, in milliseconds. ***Note:*** The JDMI default = 4992 milliseconds.

LD 73 – Change the signal values for incoming/outgoing calls.

Prompt	Response	Description
...		
FALT	...	
TIME	(0)-1920	The persistence time required before signal is accepted. ***Note:*** This value is used to implement the BLOCKING signal.

LD 73 – Change the signal values for incoming calls:

Prompt	Response	Description
...		
E SEZ(R)	ABCD	SEIZE signal.
TIME	16-(56)-1000 16-(296)-1000	Duration of pulsed time on and off, in milliseconds. The default for on is 56, and for off is 296.
E SEZA(S)	ABCD N	SEIZE ACKNOWLEDGE (answer) signal.
TIME	0-(150)-800	Delay, in milliseconds, before sending SEIZE ACKNOWLEDGE.
P WNKS(S)	ABCD N	Wink Start.
TIME	10-(220)-630	Pulse length of WNKS signal, in milliseconds.
P OPCA(R)	ABCD N	OPERATOR CALLING (receive) signal.
TIME	16-(96)-1000 16-(160)-1000	Duration of pulsed time on and off, in milliseconds. The default for on is 96, and for off is 160.
E CONN(S)	ABCD	CONNECT (answer) signal.
TIME	10-(150)-630	Pulse length of CONN signal, in milliseconds.

C CLRB(S)	ABCD/N	CLEAR BACK (answer) signal.
TIME	10-(600)-630	Pulse length of CLRB signal, in milliseconds.
P BRLS(S)	ABCD N	BACKWARD RELEASE (answer) signal.
TIME	10-(600)-2000	Pulse length of BACKWARD RELEASE signal, in milliseconds.
P FRLS(R)	ABCD N	FORWARD RELEASE (receive) signal.
TIME	16-(296)-2000 16-(960)-2000	Duration of pulsed time on and off, in milliseconds. The default for on is 296, and for off is 960.

LD 73 – Change the signal values for outgoing calls.

Prompt	Response	Description
...		
E SEZ(S)	ABCD	SEIZE signal.
TIME	10-(150)-630	Delay, in milliseconds, before sending SEIZE signal.
E SEZA(R)	ABCD N	SEIZE ACKNOWLEDGE (receive) signal.
P WNKS(R)	ABCD N	Wink Start (receive) signal.
TIME	16-(136)-504 16-(288)-504	Duration of pulsed time on and off, in milliseconds. The default for on is 136, and for off is 288.
E CONN(R)	ABCD	CONNECT (receive) signal.
TIME	16-(56)-1000 16-(296)-1000	Duration of pulsed time on and off, in milliseconds. The default for on is 56, and for off is 296.
C CLRB(R)	ABCD N	CLEAR BACK (receive) signal.

TIME	16-(296)-1000 16-(960)-1000	Duration of pulsed time on and off, in milliseconds. The default for on is 56, and for off is 296.
P FRLS(S)	ABCD N	FORWARD RELEASE (answer) signal.
TIME	10-(600)-2000	Duration of FORWARD RELEASE signal, in milliseconds.
P BRLS(R)	ABCD N	BACKWARD RELEASE (receive) signal.
TIME	16-(296)-2000 16-(960)-2000	Duration of pulsed time on and off, in milliseconds. The default for on is 296, and for off is 960.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	All
Networking:	Yes

R2 Multifrequency Compelled Signaling

R2 Multifrequency Compelled signaling is an optional software/hardware package available with Generic X11 software.

Information on R2MFC signaling and R2MFC related features can be found in the *Multifrequency Compelled Signaling* (553-2861-100) NTP.

X11 Radio Paging

The Radio Paging (RPA) feature allows radio paging equipment (radio paging system) to be connected to a Meridian 1 system. The radio paging system is a communications system used to contact mobile parties equipped with portable receivers. This communication is done via radio signals. The communication channels can be single-type (allowing one party to be paged at a time), or multiple-type (allowing several parties to be paged simultaneously).

To make a paging call, the calling party dials the paging access Flexible Feature Code. The paged party receives an indication of the incoming call in the form of a special tone, a verbal message, or a display message. The paged party can then answer the incoming call from any telephone set by dialing the answer paging Flexible Feature Code. The calling party remains off-hook until the call is answered. If all paging trunks are busy, the calling party receives a special congestion tone. The call can be tried again by redialing, or by activating the Ring Again feature.

When making a paging call, the system requires a paging access code, a mode digit, and dialed digit information. The paging access code is used by the paging system to identify the pager. The system derives this paging code by translating the DN of the party to be paged. This translation can be done in different ways, as described in this module. The mode digit indicates the type of display to be sent to the pager equipment (there are five possible display types). The digit information pertains to the calling party's DN. Depending on the type of paging chosen by the customer, this information is either entered manually by the calling party, or automatically by the system.

Local Radio Paging

To initiate a paging call, the Radio Paging System (RPS) requires the following activation sequence:

— a Paging System Access (PSA) code,

— a mode digit, and

— information digits.

The PSA code is the number used to identify a particular paging device. This code is derived by using the Directory Number (DN) of the party to be paged as a variable in the DN-PSA code translation procedure. If a valid DN is entered, the Meridian 1 sends the PSA code to the RPS that pages the party. If an invalid DN is entered, translation cannot be done and the caller receives Call To Vacant Number (CTVN) treatment. The caller can optionally page continuously until the following conditions are met:

— the paged party answers the page,

— the caller goes on-hook, and

— the paging call times out.

The paged party is required to answer the paging call within a specified time limit. When a paging call is not answered in time and the caller remains off-hook, a meet-me operation is possible. With this operation, calling parties to a radio pager are placed in a queue for a period of time, and the paged party can connect to the caller by dialing the answering Flexible Feature Code (FFC) and the paged party's DN. This connection appears as a simple call between two sets.

The paging time limits only apply to calls internal to the Meridian 1. All external calls transferred to the RPA feature will be subject to the recall timer (not the normal attendant recall) if the call is not answered.

The paged party can answer a paging call from one of the following:

— A set connected to the Meridian 1 by dialing the answering FFC followed by their own DN in order to connect to the caller and free the paging trunk.

— A Public Switched Telephone Network (PSTN) telephone in order to contact the Meridian 1 attendant and request that the paging call be answered. The attendant dials the answering FFC followed by the DN to connect to the caller while the paged party is held on the Attendant Console's source-side. The two parties are then connected in the normal way.

When there are multiple paging calls to a pager, any attempt to page a party already engaged in a paging call will receive ringback (if configured) from the Meridian 1 or call progress tones from the RPS. The caller will continue to page until the paged party answers or the caller recalls.

Remote Radio Paging

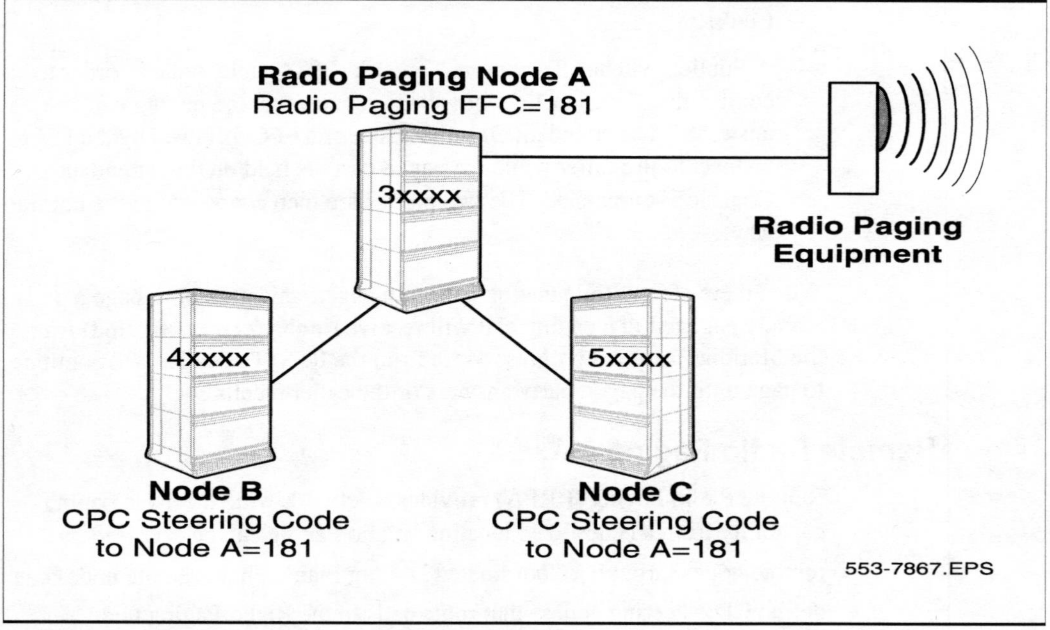
Remote Radio Paging (RRPA) provides a network-wide meet-me paging capability from a centralized location. Radio Paging can be accessed by remote nodes through a Coordinated Dialing Plan[1]. These remote nodes can define CDP steering codes[2] that route calls to the Radio Paging node.

1. The Radio Paging (RPA) package is not required at remote nodes, unless post-selection Radio Paging is required.

2. These steering codes are the equivalent of Flexible Feature Codes for Radio Paging, and are referred to as *Remote Radio Paging (RRPA) FFCs*. The steering codes must not be deleted by digit manipulation, since the digits are interpreted as the Radio Paging FFC at the radio paging node.

Figure 73 demonstrates a possible Remote Radio Paging configuration:

Figure 73
A typical Remote Radio paging configuration



Node A, which is equipped with the Remote Radio Paging feature, is referred to as the Radio Paging node. The Radio Paging FFC is defined as 181. At remote nodes B and C, steering codes of 181 have been defined to route calls to node A. To access Radio Paging from nodes B and C, a caller simply has to dial 181.

Post Selection Access to Remote Radio Paging

Remote Radio Paging allows the *post selection* operation of Radio Paging from all nodes in the network. For this functionality, all nodes must be equipped with the Remote Radio Paging feature. For post-selection access, Trunk Steering Codes (TSCs) and Distant Steering Codes (DSCs) are defined as Remote Radio Paging (RRPA) FFCs.

If a post-selection access is made to a set on the same node, the originally-called set must be either ringing or busy. If the originally-dialed set is on another node, it must be on an established call. In this latter case, the established call is disconnected before being routed to the radio paging node.

Post-selection access can be performed from PBX-type sets, SL-1 sets, Meridian 1000 series sets and Meridian digital sets, and Attendant Consoles.

Directory number to Paging System Access (PSA) code translation

Each mobile paging device is identified by a unique PSA code. A single DN can only be translated to one PSA code. The following are the different types of translation methods available:

- no translation with DN sent as PSA code (single digits can be outpulsed immediately as dialed, or batched and sent all together)
- last two digits of DN sent as PSA code
- last three digits of DN sent as PSA code
- last four digits of DN sent as PSA code or
- a translation table is searched, and the stored PSA code for the DN is sent (several DNs can be associated with a single PSA code)

With the Group Hunting feature, it is possible to forward a call to a pilot DN which points to the table containing a list of DNs to be called. In this table the RPAC and DN for RPA can be stored.

Invalid directory number handling

With the first four methods, it is not possible for the Meridian 1 to detect if the DN is invalid. With the last method, an invalid DN is blocked with the caller receiving CTVN treatment. An individual with no telephone (or DN with which to associate) can use RPA through the use of a dummy DN. The method in which an RPS responds to an invalid PSA code varies by system.

Multiple Radio Paging systems

The RPA feature allows up to 16 (numbered 0 - 15) RPSs to be configured. The following are required to configure the RPA data block:

— The translation table is to be used for all systems.

— The DN is entered with respect to a particular system number.

Paging indications

The Radio Paging Access Code (RPAC), which is a defined FFC, allows access to the procedures required to initiate a paging call. After the access FFC is dialed, the caller receives the paging tone which is removed after the first digit of the DN is entered. Seizing the trunk to the RPS before or after dialing the DN, depends on the number of RPSs configured for the customer. After the initiating FFC and DN are entered, ringback can be provided or the RPS tones can be received.

If a trunk to the RPS is not available, the caller will receive the configured congestion busy tone. The call will have to be repeated when a trunk becomes available or the Ring Again feature is used (not for an inoperative RPS). The Meridian 1 will seize an idle paging trunk and send a PSA code to the RPS.

The following are cases where a tone from the Meridian 1 will be returned to the caller to indicate that paging is in progress:

— If ringback is not required, no tone is provided (some RPSs provide call progress tones to the caller);

— If ringback and detection-of-call-accepted signal are selected, then the caller gets the ringback tone (only after receiving a call accepted signal from the RPS); and

— If ringback is required and detection-of-call-accepted signal is not required, then the caller gets the ringback tone after the valid entry of the FFC and DN.

When the caller is call forwarded (by CFNA or CFWAC) to a radio pager, a Recorded Announcement (RAN) can be sent to the caller.

Dialing plans

Two types of dialing plans can be used in a Meridian 1 network:

— **Coordinated**. A single dialing plan is created to cover all the Meridian 1s.

— **Independent**. Each Meridian 1 switch has its own dialing plan, and the Meridian 1s are connected by the use of RPACs.

The dialing plans can be arranged in various ways which can affect the way RPA works and how RPACs are manipulated.

With regard to dialing plans, the RPAC must be numeric to allow access from a second Meridian 1. Also, the Calling Line Identification (CLID) is displayed if RPA is equipped, otherwise the route access code and member number are displayed.

Single paging system

This arrangement has two or more Meridian 1s connected, but only one Meridian 1 (the source) has a RPS connected. Sets connected to any connected Meridian 1 can page any party using the same RPAC. The paged party can answer a paging call from any set on the source Meridian 1. A set on a non-source Meridian 1 can connect to a set on the source Meridian 1 by dialing the DN. If the call is redirected (for example, by ATT, CFW, or CFNA) the set on the non-source Meridian 1 can access the RPS.

Multiple paging systems

This arrangement has two or more Meridian 1s connected, with each having a connected RPS. Different RPACs are required for each RPS (the user must be aware of which Meridian 1 is connected to which RPS). Trunk access between Meridian 1s is handled by internal manipulation of the RPACs. When possible, RPSs should be connected to the same Meridian 1.

Radio Paging system signals

The RPS has two categories of signals:

State of paging call

The following are the signals an RPS can send to the Meridian 1 in order to indicate the state of the paging call:

— A disconnect signal indicates that the paging trunk can be dropped;

— A call progress signal followed by a disconnect signal indicates a paging call is in progress;

— An all-digits-received signal indicates that all required digits are received;

— An absence signal, which is the receiving of a disconnect signal before a call progress signal, indicates that a pager is installed in the paging rack. (Calls to the pager in the rack receive the congestion tone from the Meridian 1.); and

— A paging-call-accepted signal indicates that the call is accepted.

Fault-clearing and maintenance

The Meridian 1 can interpret the ready-for-service signal from an RPS. The following Meridian 1 procedures occur when a fault on the paging hardware is detected:

1 All paging calls are dropped.

2 All trunks on the faulty system are made maintenance busy.

3 Subsequent paging calls on the faulty RPS will receive maintenance-busy treatment.

The following Meridian 1 procedures occur when the fault is corrected:

1 Unbusy all trunks on a RPS.

2 Each RPS is checked (faulty systems are made maintenance busy) after a system initializes and/or reloads.

Paging time limits

For sets internal to Meridian 1 network

Each RPAC has time limits defining how long a paged party has to answer a call. (The time limits only apply to sets internal to the Meridian 1 network, as external calls are subject to attendant (ATT) recall.) The following are the three paging timers:

- **Speechpath**. For the duration, the path is maintained.
- **Non-speechpath**. For the duration, a paging trunk is held to send digit information to the RPS.
- **Meet-me**. For the period of time to perform a meet-me operation, started after outpulsing is finished (interdigit timing is used for timing the DN entry).

The paging timers can be configured in the following two ways:

- A warning tone is given eight seconds before a speechpath is dropped. After the speechpath timer expires, the trunk is dropped and the paged party is put under the meet-me timer. The caller is kept in a meet-me queue for this time.
- The paging trunk is dropped if a paging-call-accepted signal is sent by the RPS. If a paging-call-accepted signal is not sent, after the non-speechpath timer expires the paging trunk is dropped. A meet-me timer then comes into effect.

  If a paging call is not answered before the meet-me timer is activated, the paging trunk is dropped (available for other calls) and the paging device stops paging.

  If a paging call is not answered after the meet-me timer has expired, the paging set is subjected to line lock-out procedures and ringback (if configured) to the caller is stopped.

For sets external to Meridian 1 network

The recall timer overrides the existing Attendant Recall on all external calls transferred to the paging trunks. The recall timer is required because a paged party is expected to take a longer time to answer a call. Any recall to the attendant is presented to the attendant as a recall Incoming Call Indicator (ICI). Forwarded calls to the RPS will recall to the attendant. External calls are transferred to the paging equipment by the following:

— **Attendant.** Defined in the RTSA feature.

— **Set**. For calls transferred by PBX or SL-1 sets.

Methods of operation

Two different operational methods, automatic and manual, are available for RPA. Various RPACs are provided for in each method. Each RPAC has different options associated with it.

Automatic

The Meridian 1 sends all necessary digit information automatically for the caller. The digit information cannot be modified.

The following are the procedures for an RPA call:

1 Enter the RPAC.

2 Enter the DN of the paged party.

The Meridian 1 then transmits the following digit information to the RPS:

a PSA code of the receiving device,

b mode digit, and

c the DN of the caller, if required (DN key used to page call).

Manual

The caller is required to enter the mode of operation that is desired. The caller sends any required digit information from the set.

The following are the procedures for an RPA call:

1 Enter the RPAC.

2 Enter the DN of the paged party (optionally translated to a PSA code).

3 Enter the mode digit.

4 Enter the necessary digit information.

5 Enter # to indicate the end-of-digit information.

The Meridian 1 then transmits the following digit information to the RPS:

a PSA code of the receiving device

b mode digit, and

c all entered digit information.

Parallel paging

Parallel paging is a type of operation that applies to some TIE trunk interfaces (primarily used in Switzerland).

Parallel paging has the following characteristics:

— The caller remains off-hook until the paged party answers or until the call is terminated.

— The caller does not get any call progress tones from the RPS, only ringback from the Meridian 1.

— The paged party's receiving device only has the capability of indicating that there is a call.

— Only the display bleep mode of operation is allowed.

— The caller receives no indication that a PSA code is invalid. The Meridian 1 supplies ringback tone until the call times out.

Initiating a paging call

Each of the following two procedures for initiating a paging call use the same RPAC, but require that the DN be dialed at different times.

Pre-selection

Radio Paging is accessed immediately by entering the RPAC and the DN. The caller knows the RPA feature is required before going off-hook.

Post-selection

The caller dials the DN before knowing that RPA is required. While receiving ringback or busy tone, the caller dials an RPAC (an FFC) to make the destination set stop ringing (the DN of the paged party does not have to be entered a second time).

When the caller puts a call on hold (For example, by Call Transfer or Conference key) and dials another set, post-selecting on Call Transfer or Conference is not allowed.

The automatic and manual methods of operation allow post-selection access to RPA. Single-digit post-selection access codes are not supported at Remote Radio Paging (RRPA) nodes.

The following are ways to perform post-selection access to RPA:

From a PBX set

The caller sends a recall signal and receives a special dial tone, then dials the required RPAC or has single-digit access using the 16-digit post-selection feature. The caller receives Call to Vacant Number (CTVN) treatment if the RPAC is invalid.

From an SL-1 set

The caller presses the RPAG key (that has an RPAC associated with it) or 0 - 9 using single-digit post-selection to access RPA. The caller receives CTVN treatment if the RPAC is invalid.

From an Attendant Console

The caller presses the RPAG key (configured with an RPAC) to contact the paged party. The attendant receives no special dial tone, and the PAG key lamp is not used. When the RPAG key is pressed, the flashing SRC or DEST lamp becomes lit if the post-selection was successful, otherwise it remains flashing.

Modes of operation

A variety of modes, defined in mode digits, are available to allow the caller to send different types of digit information to the pager before completing the paging procedure. Some mode digits require additional information from the caller. The mode digits conform to the European Selective Paging Manufacturers Association (ESPA) standards. The caller can optionally receive call progress tones from the RPS while off-hook.

When the attendant extends a call to a pager that is in the rack, an absence signal is returned and the call is relinked into the attendant queue. When a telephone extends a call to a pager that is in the rack, the call is recalled to the set.

The following are the five mode digits:

Mode 1: External meet-me display

With Mode 1, the paged party receives a bleep and/or EXT is displayed (for external caller) on the pager. The external number or trunk route and member number are not sent by the Meridian 1. The paged party accesses a telephone and enters the answering RPAC (an FFC) followed by their DN. The Meridian 1 connects the two parties.

Mode 2: Internal meet-me display

With Mode 2, the paged party receives a bleep and/or the caller's DN (1 to 7 digits) is displayed in the form MM*dn* on the pager. The paged party accesses a telephone and enters the answering RPAC (an FFC) followed by their DN. The Meridian 1 connects the two parties. Network (ISDN) calls are considered internal and display the set's Calling Line Identification (CLID).

Mode 3: Display bleep

With Mode 3, the paged party receives a bleep and/or the caller's DN (1 to 7 digits) is displayed in the form C*dn* on the pager. The paged party makes a simple call to the caller.

Mode 4: Two-way speechpath

With mode 4, the paged party receives a bleep and the caller's DN (1 to 7 digits) is displayed on the pager. A two-way speechpath (between the caller and pager) is created for a specified period of time.

Mode 5: Alarm display

With Mode 5, the paged party receives a bleep frequency and/or unique text is displayed (explaining the urgency of the call) and/or the caller's DN. The paged party makes a call to the caller.

Note: This mode is for emergency use only.

Terminating a paging call

The Radio Paging trunk can be released in the following four ways:

— The paged party answers the paging call by dialing the answering RPAC followed by their DN.

— The caller goes on-hook.

— The paging call times out.

— A disconnect signal is sent from the RPS.

Operating Parameters

A maximum of 16 RPSs are allowed per customer.

The number of channels to the RPS is limited to the number of trunk members allowed for a trunk route.

A PSA code must be a minimum of one digit to a maximum of seven digits in length.

Single-digit post-selection access codes are not supported at Remote Radio Paging (RRPA) nodes.

Post-selection access at RRPA nodes is not supported on the ABCD keys of ABCD sets.

All DNs in the network must have the same fixed length.

The RPA feature is offered to each system disk as a package only.

The translation table size is restricted by the amount of memory available.

The serial type of paging is not supported.

The RPA feature is not available within a Dial Intercom Group (DIG).

The Multi-party Operations (MPO) Three-party Service does not work while RPA is in progress.

Call transferring an RPA call to another party is not supported.

Adding an RPA call to a conference is not supported.

Since ISDN BRI sets do not support FFCs, they cannot be used to access or answer RPA calls if the BRI sets are local on the paging node. For network situations, BRI sets can access and answer remote RPA calls. This is possible because the RPAX/RPAN FFCs are dialed as DSC/TSC steering codes.

For network RPA recall, the originating, tandem and paging nodes must be Meridian 1 switches, running a minimum of X11 Release 20 software.

QCW2/3/4 consoles are not supported for name display purposes.

For the Pre-selection to Paging situation, if the paged DN following the RPAX FFC is not local to the paging node, the CPND name for this DN cannot be obtained to display on the calling party. If the paged DN is local on the paging node and has CPND defined, the CPND can be retrieved and sent to the calling party for display purposes. For Post-selection to Paging, the CPND of the paged DN will be displayed even if the DN is not local to the paging node.

There is an existing option that allows the replacement of the RPAX FFC with a character string on set's displays. This is controlled by the DCHR prompt in LD 58. This only applies to the local paging node. On the remote node, the RPAX FFC is treated as DSC/TSC and therefore will be displayed as it is. This is an existing limitation of network Radio Paging and remains unchanged.

If a network call comes in to a set on the paging node and is redirected to paging by CFNA, the calling name cannot be retrieved and updated on the answering set when the paging call is answered. This happens only if the set on the paging node has CPND defined. If the set does not have CPND defined, the calling name could be updated on the answering party. This is a design limitation.

The following hardware is required for RPA operation:

— Televerket (TVT) Tateco system T-800 or T-900.

- Hasler system DS-1000 or DS-2000.
- NT trunk cards for parallel paging QPC*xxx* (TIE).

Feature interactions

Access restrictions

The RPA feature uses a TIE or Central Office (CO)/Public Exchange route to connect the Meridian 1 with the RPS equipment. This has some impact on current restrictions when the route is used for this purpose.

Class of Service restrictions

All restrictions that currently apply to TIE or CO routes do not apply if the route is used for Radio Paging. Any restricted set is capable of initiating an RPA call, while any set can be used to answer a paging call. The restricted set is capable of answering a paging call, even if it is from the exchange network.

Trunk Group Access Restrictions codes

The TIE or CO routes that are used for the RPA feature are subject to the limitations applied by Trunk Group Access Restrictions (TGAR) codes. Sets can be prevented from using RPA, but only after the RPAC entry. The restriction applies when accessing RPA and not when answering a call.

Trunk Barring

The normal trunk-to-trunk restrictions apply to the TIE or CO routes that are used for Radio Paging.

Attendant Recall

An RPA caller using a PBX set cannot recall the attendant by flashing, as it is ignored.

An RPA caller using an SL-1 set cannot activate the ATT recall key, as it is ignored.

Prior to X11 Release 20, the Radio Paging (RPA) feature supported Attendant Recall in standalone operation only. The RPA recalls to the local attendant on the node where the RPA system is directly connected. This product improvement enables RPA to recall the attendant who originated the Radio Paging call only; the attendant may be located anywhere within a Meridian Customer Defined Network (MCDN).

The improvement also allows the attendant's display to be updated with paged name and to display paged name instead of answering name on the paged party when answered. In addition, the improvement enables network Radio Paging to show the same display information as in standalone operation.

Automatic Call Distribution

An Automatic Call Distribution (ACD) agent is allowed to transfer a call to RPA. The following are the operations:

- When a recall takes place and the transferring party is an ACD agent, the call is recalled to the ACD queue.
- When an RPA call is answered before the recall is presented to an ACD agent, the recall is removed from the queue.
- When an RPA call is answered while recall is presented to an ACD agent, the ringing is removed and the ACD agent is idled for other calls.
- When an RPA call is dropped while recall is presented to an ACD agent, it appears to the ACD agent as if the call was answered.
- When an ACD agent with an RPA recall presented presses a DN or a Make Set Busy key, the recall is removed from that ACD agent and a new recall to the ACD agent is attempted. If no ACD agents exists, the call is recalled to the attendant.

Note: It is not possible to answer an RPA call that has recalled to an ACD agent with the Call Force option.

Automatic Dialing

The Autodial key can be programmed to perform RPA.

Automatic Timed Reminders

A new RPA recall timer (longer duration) overrides the existing recall timer. This RPA recall timer applies only to Public Switched Telephone Network (PSTN) and direct inward dialing (DID) sets using RPA trunks. The call receives Recall To Same Attendant (RTSA) treatment if the paging call is not answered by the paged party within the specified time.

Barge-in

Barge-in to either a caller trunk or an RPA trunk, while RPA is in operation, is not permitted and results in an overflow tone being returned to the attendant. The RPA operation is not affected and the paging will continue until one of the following occurs:

— the caller goes on-hook;

— the call is answered; or

— the call times out.

If an attendant attempts to Barge-in to an RPA trunk that is not busy, the trunk is seized and a dial tone is returned to the attendant. The attendant can then dial a PSA code to page the desired party. The method of operation is the same as Barge-in to an idle trunk.

Basic Automatic Route Selection

Radio Paging CO and TIE trunk routes can be set up with BARS.

> ***Note:*** These routes should not be entered in a schedule with normal CO or TIE routes, because they will respond differently.

Break-in

Break-in to either a caller or paged party, while RPA is in operation, is not permitted and results in an overflow tone being returned to the attendant. The RPA operation is not affected, and paging continues until one of the following occurs:

— the caller goes on-hook;

— the call is answered; or

— the call times out.

Busy Verify

Busy Verify for either a caller or paged party, while RPA is in operation, is not permitted and results in an overflow tone being returned to the attendant. The RPA operation is not affected and the paging will continue until one of the following occurs:

— the caller goes on-hook;

— the call is answered; or

— the call times out.

Call Detail Recording

Call Detail Recording (CDR) has two types of operation:

CDR on incoming or outgoing calls to Radio Paging system

In the first type, no CDR S record (between trunk and transferred party) is printed until the call is answered. Upon disconnection of an answered paging call, a CDR E record (between trunk and paged party) is printed, identifying the paged party DN and not the DN of the set from which the call was answered. Call Detail Recording (CDR) for internal calls is consistent with CDR for external calls.

No CDR record is printed on paging recalls which are re-extended to the paging trunk.

CDR on paging route

With this X11 Release 20 enhancement to CDR, an “S” record is printed when an attendant extends an outgoing trunk call to a destination party. When the extended outgoing trunk call or the destination party releases to disconnect, an “E” record is printed. Prior to this enhancement, when a call accesses the RPA route with CDR, a CDR N record (between caller and the paging trunk) is printed after the call is answered, abandoned or timed out.

Call Forward

Call Forward All Calls

This feature can allow equipped PBX or SL-1 sets to have calls automatically forwarded to an RPAC. This forwarded number can be numeric or a non-numeric version in the FFC table.

Forwarding internal and external calls to the RPS requires the call forwarding number be defined as the RPAC and DN of paging device. If just the RPAC is entered, the paging DN is that of the set where CFW is activated. The RPS can provide a RAN for the caller.

Call Forward No Answer

A call to a PBX or SL-1 set that is not answered after a specific number of rings is automatically forwarded to an RPS.

Call Transfer

A call can be transferred to an RPS with the following conditions: internal calls are subject to paging time outs; and external calls are subject to recall.

When transferring a call to an RPS, the transferring party may use pre-selection or post-selection method of access.

Call transferring an RPA call to another party is not supported.

Central Office/Public Exchange trunks

Central Office/Public Exchange trunks can be used for transfer of information to an RPS when the call progress tones from the RPS are received.

Conference

While in a conference, a party can make a paging call by using one of the following: switchhook flash (from an SL-1 set), Transfer (TRN) key, or Conference (A06) key (from a BCS set).

When the RPA call is complete, the party can drop Radio Paging and return to the conference. Adding an RPA call to a conference is not supported.

Dial 1

Using the register recall on a PBX set, while receiving ringback tone, is allowed. If register recall is not allowed for a user, a ground button is used to allow post-selection initiation.

Digit Display

SL-1 set

During RPA operation, the display shows the FFC and DN for pre-selection and the DN FFC for post-selection initiation. When a call is re-routed (forwarded, hunted or transferred) to the RPS, the caller's display shows the FFC and paged party DN. After a paging call is answered, the caller's display is updated to show the answering set's DN. The paged party's set displays the caller's DN.

Attendant Console

The display is similar to the SL-1 set when accessing and answering RPA calls. When a recall from paging occurs, the Attendant Console display shows the RPA FFC and the paged party's DN. The recall ICI key also indicates that the paging call has recalled.

The CLID is displayed if that feature is equipped. With CPND, the paged party's name supplements their DN display. If the Display Characters (DCHR) option is used in the RPA (LD 58), the FFC DN is replaced by the specified characters.

Direct Inward Dialing

When an incoming DID trunk attempts to gain access to a TIE or CO trunk that is configured as having RPS equipment, these calls are not intercepted by the attendant. The RPA call is made in the normal manner. The RPAC must be numeric.

Direct Inward System Access

Public Switched Telephone Network (PSTN) calls, accessing the RPA trunk, are handled in the same fashion as direct inward dialing calls.

Do Not Disturb

A set (DN) in the Do Not Disturb (DND) state can receive paging calls.

Enhanced Flexible Hotline

The RPAC (FFC) and DN can be stored in a hotline list of pre-set digits.

Group Hunting

With Group Hunting, it is possible to forward a call to a pilot DN that points to a table containing a list of DNs to be called. In this Group Hunting table, the RPAC (FFC) and DN for RPA can be stored.

Hold

The Hold key or autohold works on a paging call as if a station-to-station call is being made. The caller's set can be on hold while receiving a ringback tone or call progress tones. When a paging call is put on hold, no indication is given if the call has been answered. The Attendant Console SRC lamp is continuously lit, from the winking state, when the call that is put on hold is answered.

Last Number Redial

When a valid RPA FFC with a DN is entered and the configured length is enough, the FFC and DN are stored. When a manual RPA FFC is entered, the information digits and octothorpe (#) character are also stored.

Multifrequency Compelled Signaling (MFC)

Radio Paging can be accessed by a diversion from TIE or DID trunks using MFC.

The idle signal is not sent immediately when the RPA trunk is seized, since the RPS answers with a call accepted signal or a busy signal (when the ACPS prompt is set to YES). An idle signal is sent back immediately when one of the following occurs:

— no signal can be sent back from the RPS (when the ACPS prompt is set to NO);

— a Recorded Announcement (RAN) is provided; or

— Recall on Busy is configured.

Multiple Appearance Directory Number

With a Multiple Appearance DN, only one receiving device PSA code can be associated with the DN (not associated with a particular set).

Multiple Customer Operation

Each customer can connect to the RPS equipment. The RPSs connected are independent of each other.

Multi-party Operations (MPO)

It is possible to hold an existing call (during Call Join, Three-party Service or Conference-6) and initiate or answer a paging call. Transferring an external call is subject to the RPA Recall timer. When there is no answer to an initiated paging call, the call is released in the normal manner by pressing the DN key again on an SL-1 set or pressing Register Recall on a PBX set. The MPO user can toggle between an established call and a paging call.

> *Note:* Three-party Service does not work while RPA is in progress. If the caller flashes with an established held call and an active unanswered paging call, the paging call is stopped and the held call is reestablished as active.

Network Automatic Route Selection (NARS)

Radio Paging CO and TIE trunk routes can be set up with NARS.

Note: These routes should not be entered in a schedule with normal CO or TIE routes because they will respond differently.

Night Service

Incoming calls to a Night Service set (DN) can be transferred to RPA DNs. Calls can be entered or answered from the Night Service set. External calls transferred to RPA DNs recall to the Night Service DN.

Override

This feature allows a set to break into an existing call. The Break-in feature restrictions apply.

Ring Again

The RPA feature allows Ring Again to be applied when a paging route is busy. The caller can re-apply Ring Again when the congestion tone is received.

With RPA post-selection access and a caller attempting Ring Again, the indications that Ring Again is already activated or the queue is too large cannot be given until the RPAC has been dialed.

With RPA pre-selection access to a single RPS, the busy trunk indication is given immediately after the RPAC (FFC) is dialed. Ring Again only redials the trunk (on SL-1 sets all digits entered after the busy tone are redialed). The DN to be paged has to be re-entered.

With RPA pre-selection access to multiple RPSs and RPA post-selection access to a single RPS or multiple RPSs, the busy trunk indication is given after the DN is entered. Ring Again redials the trunk and the DN (all digit information in the automatic method is also redialed). Ring Again is ignored when a set is forwarded to the RPS, and all the trunks are busy.

Slow Answer Recall

A paging call is recalled to the attendant if it has gone unanswered after a period of time. The attendant uses the RLS key to extend the call again. The attendant console displays the RPAC (FFC), DN and CLID when there is a recall from paging.

Slow Answer Recall Modification (SLAM)

With the Slow Answer Recall Modification feature enabled, when the attendant answers a recall, the destination party is disconnected.

When the attendant answers a paging recall, the call is removed from the meet-me queue and the recall cannot be answered by the paging party by using RPA Answer. The paging party is put on the source side of the attendant; there is nothing connected on the destination side. The attendant cannot extend the call to paging by pressing the Release key. Pressing the Release key will disconnect the paging party from the source side and the attendant will become idle.

The attendant can extend the call to Radio Paging again by either: dialing the RPAX FFC + the DN (preselection); or dialing the DN, and while the DN is ringing or busy pressing the RPAG key (post-selection).

Speed Call

The Speed Call feature can be set up to perform RPA dialing.

Station-to-station calling

When a party is paged by one caller and a second party dials the paged party's DN, the call will ring the paged party's set in the normal manner.

Switchhook Flash

Using the register recall on a PBX set is allowed while receiving a Ringback tone. If register recall is not allowed for a user, a ground (earth) button is used to allow the post-selection access method.

Tenant Service

A tenant can be restricted from accessing an RPA trunk and can be configured to share or privately use an RPA trunk. All other restrictions apply to RPA.

TIE trunks

This trunk type is used for information transfer to an RPS. Special hardware is required.

Traffic Measurements

The following traffic measurements are available for RPA:

— **Paging recall count.** Incremented each time a paging call is recalled to the attendant.

— **Average answer time.** The average time paging calls are in the paging queue before being answered.

Trunk Group Busy Indication

The Attendant Console's Trunk Group Busy (TGB) key/lamp pair can be assigned to each of the RPA trunk routes. The attendant presses the TGB key to deny a set access to a RPS. The TGB lamp goes on and all calls to the RPS are routed automatically to the attendant. Normal RPS access returns and the lamp goes off when the attendant presses the TGB key again. The following conditions apply to sets with TGAR:

— Sets with TGAR of 0 to 7 are routed to the attendant if the trunk group being accessed has been made busy by the attendant.

— Sets with TGAR of 8 to 15 are not restricted by the TGB operation by the attendant.

The TGB lamp flashes when all trunks in the paging trunk group are busy.

When a RPS is faulty, its TGB lamp flashes after all associated (with the faulty paging route) trunks have been made maintenance-busy. The reverse happens when the fault is corrected in the RPS hardware.

Feature packaging

The following feature packages are required for paging operation in addition to the Radio Paging (RPA) package 187:

— Flexible Feature Codes (FFC) package 139 (to gain access to RPA);

— 16-Button Dual-tone Multifrequency Telephone (ABCD) package 144 (to allow single digit post-selection access to RPA);

— For the Radio Paging network recall operation, Network Attendant Service (NAS) package 159 must be provisioned;

— For Remote Radio Paging, Coordinated Dialing Plan (CDP) package 59 is required to define RPA FFCs as Distant Steering Codes (DSCs) or Trunk Steering Codes (TSCs);

— To display characters instead of the Radio Paging Flexible Feature Code, Calling Party Name Display (CPND) package 95 is required; and

— Integrated Services Digital Network (ISDN) package 145 and its dependencies are required for operation in a Meridian Customer Defined (MCDN) ISDN network.

Feature implementation

Adding a Radio Paging System

LD 15 – Enable or disable the RPA feature.

Prompt	Response	Description
REQ	CHG	Change existing data block.
TYPE	CDB FTR	Customer Data Block. Feature and options data. Release 21 gate opener.
CUST	0-99 0-31	Customer number. For Option 11C.
RPA	(NO) YES	Radio Paging Allowed.

LD 16 – Configure trunk route for Radio Paging feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	0-99 0-31	Customer number, as defined in LD 15. For Option 11C.
ROUT	0-511 0-127	Route Number. For Option 11C.
TKTP	TIE COT	Trunk route.

RPA	(NO) YES	Radio Paging Route.
OPR	(YES) NO	Outpulsing Route (YES is the default if RPA = YES).

LD 14 – Enable or disable the reversing of the E-lead.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	TIE COT	TIE trunk. Central Office trunk.
TN	l s c u c u	Terminal Number For Option 11.
CUST	xx	Customer Number.
CLS	RVEP XREP	Reverse earpiece. Do not reverse earpiece.

LD 11 – Configure the RPAG key for SL-1 sets.

Prompt	Response	Description
REQ	CHG	Change RPAG key assignment.
TYPE	aaaa	Telephone type, where xxxx = SL1, 2006, 2008, 2009, 2016, 2112, 2216, 2317, 2616 or 3000
TN	l s c u c u	Terminal Number. For Option 11.
KEY	xx RPAG yyyy	To define an RPAG key with an RPAC (FFC), where xx is a key number and yyyy is an RPAC.

LD 12 – Configure the RPAG key for Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	xxxx	Type of Attendant Console. xxxx = 1250, 2250.
TN	l s c u c u	Terminal Number. For Option 11.
KEY	xx RPAG yyyy	To define an RPAG key with an RPAC (FFC), where xx is the key number and yyyy is an RPAC.

LD 56 – Configure the RPA warning tone.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	FTC	Flexible Tone and Ringing data block.
TABL	0-31	FTC Table Number.
SCCT	(NO) YES	Modify Software Controlled Cadences and Tones.
RPAW	x xx xx xx	Radio Paging Warning tone definition.

LD 57 – Define the Flexible Feature Codes (RPACs).

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	FFC	Flexible Feature Codes Data Block.
CUST	xx	Customer Number.
CODE	RPAX	Radio Paging Access Code.

-RPAX	RPAX xxxx	Radio Paging Access Code. Enter Flexible Feature Code. The RPACs entered here are associated with various options in LD 58.
CODE	RPAN	Radio Paging Answer call code.
-RPAN	RPAN xxxx	Radio Paging Answer call code. Enter Flexible Feature Code.

LD 58 – Define RPA customer information.

Prompt	**Response**	**Description**
REQ	NEW CHG	New, or change.
TYPE	RPCD	Radio Paging Customer Data Block.
CUST	xx	Customer Number.
RPTO		Radio Paging Tone.
	SPCL DIAL NONE	Special Dialtone. Normal Dial tone. No Tone. This Radio Paging tone is provided after the RPAX and RPAN.
MRPS	(NO) YES	Multiple Radio Paging Systems.
TRAN		Translation type.
	TAB TWO THR FOR NO	Table Search. Last two digits of DN. Last three digits of DN. Last four digits of DN. None. Prompt is not given when MRPS = YES and TRAN is forced to TAB.
DNLN	1-(4)-7 (If TRAN = NO, TWO, THR or FOR)	DN length.

RCRG	0-(6)-20 X	Number of ring cycles when recall to transferring set, before reroute to attendant. (0 is the CFNA prompt value.) Reroute to attendant.
RCTI	0-(30)-120	Time to wait for a "BUSY" transferring set to become idle.
RCAL	(NO) YES	Recall if busy from RPA.
TBTR	4-(10)-30	Time between two recall attempts (to an SL-1 set).

LD 58 – Define RPS information.

Prompt	**Response**	**Description**
REQ	NEW CHG	New, or change.
TYPE	RPS	Radio Paging System Data Block.
CUST	xx	Customer Number.
SNUM	0-15	System Number.
PSAL	1-7	Paging System Access code length.
RTIM	0-(60)-630	Length of the Recall Timer.
STO	10-(30)-630	Length of time for Screech Path to be maintained in seconds.
NSTO	10-(30)-630	Length of time required for paging when no Screech Path is required.
MTO	0-(150)-630	Length of the Meet-Me Time-out timer in seconds.

LD 58 – Define the RPAC information.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RPAX	RPAC Data Block.
CUST	xx	Customer Number.
SNUM	0-15	System Number.
RPAX	nnnn	Radio Paging Access Code.
-ROUT	0-511 0-127	Route Number. For Option 11C.
-PANN	(NO) YES	Record Paging Announcement.
--RPAR	0-511 0-127	Route Number that provides the recorded announcement. For Option 11C.
-BYPS	(NO) YES	Bypass the DN-PSA translation. If BYPS = YES, then meet-me is not available, and the trunk is accessed directly.
--OPER	(AUTO) MANU	Automatic Operation. Manual Operation.
--EXTM	(0)-9 (If OPER = AUTO)	
--INTM	(0)-1-9	Internal Mode digit for this RPAX.
--TRDN	(0)-7 (If OPER = YES)	Transmit this number of digits of the caller's DN to the paging equipment.
-PATH	NONE SPCH RNGB	Speech Path or Ringback Speech Path. Ringback to the caller.
--TWSP	If PATH = SPCH (BOTH) EXT	Two-way Screech Path with a mobile pager allowed. Internal and external calls. External calls.
--ACPS	If PATH = SPCH (YES) NO	 Radio Paging System to provide the call-in-progress signals.

--ACPT	If PATH = SPCH or RNGB, (YES) NO	Call Accepted is to be detected. When PATH = RNGB or SPCH, and ACPT = YES, Ringback is provided only when the call-accepted signal is received. Speech Path opens when the start-talk signal is received. When PATH = RNGB and ACPT = NO, Ringback is provided when all the paging digit information has been sent (ending # processed). When PATH = SPCH and ACPT = NO, Speech Path is provided when all of the paging digit information has been sent (ending # processed).
--DCHR	xxxx X	Display characters. Remove all characters.

LD 58 – Change the Translation Table Information.

Prompt	**Response**	**Description**
REQ	CHG	Change.
TYPE	TBL	Translation Table access.
SNUM	0-15	System Number.
DNPS	xxxx yyyy	The DN to be translated and the number of the paging equipment to which the DN is assigned.
TABT	aaa	Table Type (Prompted when REQ = PRT)
RANG	xxxx...xxxx	Print DN Range from the first DN to the second DN (Prompted when REQ = PRT).

LD 18 – Define the ABCD table.

Prompt	Response	Description
REQ	NEW CHG	Add, or change 16 Button Data Block.
TYPE	ABCD	16 Button Data Block.
TBNO	1-254	Table Number.
DFLT	1-254	Default function table number.
PRED	(NO) YES	Pre-dial.
POST	(NO) YES	Post-dial.
CONT	(NO) YES	Control.

LD 18 – Define the pre-translation and post-translation list numbers.

Prompt	Response	Description
REQ	NEW CHG	Add, or change Pretranslation table assignment.
TYPE	PRE	Pretranslation calling group assignment.
CUST	xx	Customer Number.
XLAT	0-254 0-8191	Pretranslation list (Calling group to Speed Call list correlation.)
	0-254 8191	If list number 8191 is assigned to a group, pretranslation is removed for that group.
-PRE	0-8190 X	Pre-translation Speed Call List number. Remove list.
-PST	0-8190 X	Post-translation Speed Call List number. Remove list.
-SDA	0-8190 X	Single-digit Access Speed Call List Number. Remove list.

Adding a Remote Radio Paging Flexible Feature Code

LD 87 – Define a Remote radio Paging (RRPA) FFC.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
CUST	0-99 0-31	Customer number, as defined in LD 15. For Option 11C.
FEAT	CDP	Coordinated Dialing Plan Feature.
TYPE	DSC TSC	Distant Steering Code. Trunk Steering Code.
DSC	xxxx	Distant Steering Code.
-FLEN	(0)-10	Flexible Length number of digits.
-DSP	LSC LOC DN	Display.
-RRPA	(NO) YES	Remote Radio Paging Access.
-RLI	xxx	Route List to be accessed for distant steering code.
-CCBA	(NO) YES	Collect Call Blocking.
TSC	xxxx	Trunk Steering Code.
-FLEN	(0)-16	Flexible Length number of digits.
-ITOH	(NO) YES	Inhibit Time Out option.
-CCBA	(NO) YES	Collect Call Blocking.
-RLI	xxx	Route List to be accessed for trunk steering code.

LD 11 – Configure the RPAG key for SL-1 sets.

Prompt	Response	Description
REQ	CHG	Change RPAG key assignment.
TYPE	aaa	Type of Data Block.
TN	l s c u c u	Terminal Number. For Option 11.
KEY	xx aaa yyyy	To define an RPAG key with the RRPA FFC.

LD 12 – Configure the RPAG key for Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	xxxx	Type of Attendant Console. xxxx = 1250, 2250
TN	l s c u c u	Terminal Number. For Option 11.
KEY	xx RPAG yyyy	To define an RPAG key with the RRPA FFC.

Feature operation

The following occurs when more than one RPS is configured per customer:

— The system number is transparent to the caller;

— The DN-PSA code translation table decides which RPS to use; and

— The trunk search is done after the DN is entered.

When one RPS is configured per customer, the trunk search is made after the FFC is entered.

Different call progress tones are provided by the RPS depending on the mode digit and state of the paging call.

Automatic pre-selection

Meridian 1 proprietary or SL-1 set

The following are the operation steps:

1 Off-hook.

— Set receives dial tone.

2 Enter the RPAC (FFC) for initiating RPA.

— Set receives paging tone if FFC is valid.

— Set receives CTVN treatment if FFC is invalid.

— Set receives congestion tone (as configured) if no trunk is available in a single system.

3 Enter the DN of party to be paged.

— Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.

— Set receives no tone from the Meridian 1 if speechpath is provided.

— Set receives CTVN treatment if DN is invalid.

— Set receives congestion tone if no paging trunks are available.

— Set receives busy tone if absence signal is received.

Attendant Console

When paging from a PSTN set, the attendant can access the RPA feature using the above steps and then transfer the call (similar to transferring to a normal set).

Automatic post-selection

Single-digit post-selection access codes are not supported at Remote Radio Paging (RRPA) nodes.

Meridian 1 proprietary sets

The following are the operation steps:

1 Off-hook.

— Set receives dial tone.

2 Enter the DN of party desired to be reached.

— Set receives ringback or busy tone if DN is valid.

— Set receives CTVN treatment if DN is invalid.

3 Press Recall key.

— Set receives recall signal.

4 Press single digit 0 - 9 for speed call list.

5 Press single alphabetic A - D, where character is a RPAG key (for RPA) for 16-Button DTMF set.

6 Enter RPAC (FFC) for initiating RPA.

— Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.

— Set receives no tone from the Meridian 1 if speechpath is provided.

— Set receives CTVN treatment if FFC or DN is invalid.

— Set receives congestion tone if no paging trunks are available.

— Set receives busy tone if absence signal is received.

SL-1 set

The following are the operation steps:

1 Off-hook.

— Set receives dial tone.

2 Enter the DN of party to be paged.

— Set receives ringback or busy tone if DN is valid.

— Set receives CTVN treatment if DN is invalid.

3 While in ringing or busy state, press RPAG key (for RPA).

4 Press single digit 0 - 9 for speed call list.

— Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.

— Set receives no tone from the Meridian 1 if speechpath is provided.

— Set receives CTVN treatment if FFC or DN is invalid.

— Set receives congestion tone if no paging trunks are available.

— Set receives busy tone if absence signal is received.

Attendant Console

The following are the operation steps:

1 Off-hook.

— Set receives dial tone.

2 Enter the DN of party to be paged.

— Set receives ringback or busy tone if DN is valid.

— Set receives CTVN treatment if DN is invalid.

3 Press RPAG key (for RPA).

— Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.

— If the paging call recalls, the attendant can re-extend the call.

— Set receives CTVN treatment if FFC or DN is invalid.

— Set receives congestion tone if no paging trunks are available.

— Set receives busy tone if absence signal is received.

Manual pre-selection

Meridian 1 proprietary and SL-1 sets

The following are the operation steps:

1 Off-hook.

— Set receives dial tone.

2 Enter the RPAC (FFC) for initiating RPA.

— Set receives paging tone if FFC is valid.

— Set receives CTVN treatment if FFC is invalid.

— Set receives congestion tone (as configured) if no paging trunk is available.

3 Enter the DN of party desired to be reached.

— Set receives ringback or busy tone if DN is valid.

— Set receives CTVN treatment if DN is invalid.

4 Enter mode digit.

5 Enter information to be sent.

6 Enter # for end of information.

— Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.

— Set receives busy tone if absence signal is received.

Attendant Console

When paging from a PSTN set, the attendant can access the RPA feature using the above steps and then transfer the call (similar to transferring to a normal set).

Manual post-selection

Single-digit post-selection access codes are not supported at Remote Radio Paging (RRPA) nodes.

Meridian 1 proprietary set

The following are the operation steps:

1 Off-hook.

— Set receives dial tone.

2 Enter the DN of party to be paged.

— Set receives ringback or busy tone if DN is valid.

— Set receives CTVN treatment if DN is invalid.

3 Press Recall key.

— Set receives recall signal.

4 Press single digit 0 - 9 for speed call list.

5 Press single alphabetic A - D, where character is a RPAG key (for RPA) for 16-Button DTMF set.

6 Enter RPAC (FFC) for initiating RPA.

— Set receives no tone from the Meridian 1 if speechpath is provided.

— Set receives CTVN treatment if FFC or DN is invalid.

— Set receives congestion tone if no paging trunks are available.

7 Enter mode digit.

8 Enter information to be sent.

9 Enter # for end of information.

— Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.

— Set receives busy tone if absence signal is received.

SL-1 set

The following are the operation steps:

1 Off-hook.

— Set receives dial tone.

2 Enter the DN of party to be paged.

— Set receives ringback or busy tone if DN is valid.

— Set receives CTVN treatment if DN is invalid.

3 While in ringing or busy state, press RPAG key (for RPA).

4 Press single digit 0 - 9 for speed call list.

— Set receives no tone from the Meridian 1 if speechpath is provided.

— Set receives CTVN treatment if FFC or DN is invalid.

— Set receives congestion tone if no paging trunks are available.

5 Enter mode digit.

6 Enter information to be sent.

7 Enter # for end of information.

— Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.

— Set receives busy tone if absence signal is received.

Attendant Console

The following are the operation steps:

1 Off-hook.

— Set receives dial tone.

2 Enter the DN of party to be paged.

— Set receives ringback or busy tone if DN is valid.

— Set receives CTVN treatment if DN is invalid.

3 Press RPAG key.

— Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.

— If the paging call recalls, the attendant can re-extend the call.

— Set receives CTVN treatment if FFC or DN is invalid.

— Set receives congestion tone if no paging trunks are available.

Answering the paging call

Paged party

The paged party receives a paging indication followed by one of the following types of information:

— no information

— a short speech cut-through, or

— digits displayed on receiving device.

A paged party can respond after receiving the information, as in the following:

— When the information is the caller's DN, the paged party responds by initiating a normal station-to-station call.

— When the information is not telephone related, the receiving device might get a coded message to perform some action.

Pre-selection and post-selection

Meridian 1 proprietary and SL-1 sets

The following are the operation steps:

1 Off-hook from any set on the system.

— Set receives dial tone.

2 Enter the FFC for answering paging calls.

— Set receives paging tone if the FFC is valid.

— Set receives CTVN treatment if FFC is invalid.

3 Enter DN of your set.

— Set is connected to the caller if the DN is valid.

— Set receives CTVN treatment if the DN is invalid or is not being paged.

Attendant Console

When answering a paging call from a PSTN set, the attendant is required to make the connection. The attendant dials using the above method (FFC and DN) as if the call is being extended to another set.

Introduced in X11 Release:	20.1x
Networking:	Yes

Radio Paging Product Improvements

A Radio Paging system is a communications tool used to contact mobile parties by means of radio signals. A caller can use their telephone set to page a mobile party who has a mobile portable receiving device.

Prior to Release 20, the Radio Paging (RPA) feature supported attendant recall in standalone operation only. The RPA recalls to the local attendant on the node where the RPA system is directly connected. This product improvement enables RPA to recall the attendant who originated the Radio Paging call only; the attendant may be located anywhere within an ISDN Meridian Customer Defined Network (MCDN) configured with Network Attendant Services (NAS).

The improvement also enables an attendant's display to display paged name, instead of answering name, on the paging party when answered, and to make network Radio Paging show the same display information as in the standalone operation. For more information about Radio Paging, please see the Radio Paging feature module in this guide.

Operating parameters

Since ISDN Basic Rate Interface (BRI) sets do not support Flexible Feature Codes (FFCs), they cannot be used to access or answer RPA calls if the BRI sets are local on the paging node. For network situations, BRI sets can access and answer remote RPA calls. This is possible because the Radio Paging Access Code (RPAX)/Radio Paging Answering Code (RPAN) FFCs are dialed as Distant Steering Codes (DSCs)/Trunk Steering Codes (TSCs).

The originating, tandem and paging nodes must be Meridian 1 systems running a minimum of Release 20 software.

For Pre-selection Paging, if the paged DN following the RPAX FFC is not local to the paging node, the Call Party Name Display (CPND) name for this DN cannot be obtained to be displayed on the calling party's terminal. If the paged DN is local on the paging node and has CPND defined, the CPND can be retrieved and sent to the calling party for display purposes. For Post-selection Paging, the CPND of the paged DN will be displayed even if the DN is not local to the paging node.

If a network call comes in to a set on the paging node and is redirected to paging by Call Forward No Answer (CFNA), the calling name cannot be retrieved and updated on the answering set when the paging call is answered. This happens only if the set on the paging node has CPND defined. If the set does not have CPND defined, the calling name can updated on the answering party's set.

The following hardware is required for Radio Paging operation: Radio Paging System equipment meeting European Selective Paging Manufacturers' Association (ESPA) requirements; trunk cards (QPC296/QPC287/QPC551/QPC71/QPC237/NTD9742A/NT5K19AA) or Extended Flexible E&M (XFEM) cards (NT5K83/NT5K72/NT5K50/NT5K19).

The following hardware is required for non-Option 11 systems: PRI – QPC720; PRI2 – NT8D72; and DCH – QPC757, NT6D11, or NT6D70 (MSDL).

The following hardware is required for Option 11 systems: PRI – NTAK09 with NTAK93 data port; PRI2 – NTAK79, or NTDK50 with NTBK51 DCHI data port; ISL – NTAK02.

Feature interactions

Call Detail Recording Enhancement

When an attendant makes an outgoing call (established on the source side) and then extends the call to remote radio paging on another node by using a normal trunk (for example, Trunk X), an "S" record is printed when the attendant releases to extend the call to network RPA.

If the outgoing trunk call releases before the paged call is answered, the "E" record will show the normal trunk ID (Trunk X).

If the paged call has been answered when the outgoing trunk call releases, the "E" record will show the paged DN instead of Trunk X.

Display of Calling Party Denied

If this feature is enabled (packaged under the International Supplementary features package 131), additional Classes of Service can be assigned to sets to determine whether or not their DN and CPND information will be displayed on other sets. No CPND or DN information is displayed on sets involved in a network RPA call that have name display denied or digit display denied Class of Service.

Network Attendant Services

Network Attendant Services (NAS) configuration is a requirement for the Network Radio Paging (NRPA) Recall to Same Attendant (RTSA) feature. Without NAS, NRPA RTSA is not active, and existing operation will be followed.

With NAS configured, if an RPA recall to the attendant on the originating node is not allowed, the recall will be presented on the paging node. Existing operation prior to this development is performed. There is no new interaction introduced with NAS features.

Slow Answer Recall Modification

With the Slow Answer Recall Modification (SLAM) feature enabled, when the attendant answers a recall the destination party is disconnected. This also applies to Radio Paging.

When the attendant answers a paging recall, the call is removed from the meet-me queue and the recall cannot be answered by the paging party by using RPA Answer. The paging party is put on the source side of the attendant; there is nothing connected on the destination side. The attendant cannot extend the call to paging by pressing the Release key. Pressing the Release key will disconnect the paging party from the source side and the attendant will become idle.

The attendant can extend the call to Radio Paging again by either: dialing the RPAX FFC + the DN (preselection); or dialing the DN, and while the DN is ringing or busy pressing the RPAG key (post-selection).

Feature packaging

Radio Paging (RPA) package 187 must be provisioned to activate this feature.

To gain access to RPA, Flexible Feature Codes (FFC) package 139 must be provisioned.

For the Radio Paging network recall operation, Network Attendant Service (NAS) package 159 must be provisioned.

For Remote Radio Paging, Coordinated Dialing Plan (CDP) package 59 is required to define RPA FFCs as Distant Steering Codes (DSCs) or Trunk Steering Codes (TSCs).

To display characters instead of the Radio Paging Flexible Feature Code, Calling Party Name Display (CPND) package 95 is required.

Integrated Services Digital Network (ISDN) package 145, and its dependencies, are required for operation in an MCDN ISDN network.

Feature implementation

LD 87 – Set up remote Radio Paging on originating node.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	TSC DSC	Trunk/Distant Steering Code (enter RPAX/RPAN FFC defined on paging node).
TSC, DSC	xxxx	Radio Paging FFC from paging node.
RRPA	(NO) YES	Remote Radio Paging option.
RLI		Route List Index of route list block used to route to paging node.

LD 15 – In order for the Recall to Same Attendant portion of this feature to operate network wide, the Recall to Same Attendant (RTSA) prompt has to be activated on the originating node as follows:

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	CDB ATT	Customer Data Block. Release 21 gate opener.
...		
- RTSA	(RSAD) RSAA RSAX	Recall to same attendant denied. Recall to same attendant allowed. Recall to same attendant with queuing on busy.

Feature operation

With ISDN NAS enabled, the RPA Recall will recall to the same attendant who originated the call. The attendant may be located anywhere in the ISDN NAS network.

When the originating attendant answers the RPA recall, the call can be extended again by simply pressing the Release key.

When the paged party answers, recall to the originating attendant will be cancelled if the attendant has not yet answered.

If the paged party answers while the paging call is recalled to the originating attendant (buzzing), the request to cancel the recall is sent from the paging node to the originating node. If the attendant answers the recall before receiving the cancel message, the attendant is connected to both the paging and answering parties.

If the RPA RTSA network wide feature is not allowed, the recall is presented on the paging node. Existing operation prior to this development is performed. The RPA RTSA network wide feature is not allowed when one of the following conditions occurs:

— The originating attendant is busy (active on a loop) and RTSA is not RSAX on the originating node.

- — The originating attendant is disabled or in maintenance mode.
- — The originating attendant is in Night Service.
- — The originating attendant is in Position Busy mode.
- — The paging call was not handled by an attendant on the originating node. This includes:
  - A set directly dials access to remote paging.
  - The call is transferred by a set to remote paging.
  - An attendant dials access to remote paging on the source side, with no other parties involved.
- — The originating attendant never released to extend the paging call to the calling party (i.e., the attendant has the calling set on the source side and the paging call on the destination side at recall time).

The recall time out for an RPA call is defined on the node that is directly connected to the RPA system, not the originating node from where the attendant extended the call. This is because the RPA timer is usually longer than the normal recall time out so that the paged party will have enough time to answer the call.

Introduced in X11 Release:	23.0x
Networking:	Yes

Radio Paging Product Improvement Continuation

A Radio Paging System (RPS) is a communications tool used to contact mobile parties by means of radio signals. With this system, a set can page a mobile party that is equipped with a radio paging device. The Radio Paging Product Improvement Continuation enhances the performance of the Radio Paging feature by providing the following:

— an increase in the number of digits sent to and displayed on a Radio Paging device

— the ability to activate/deactivate Pretranslation for Radio Paging calls

— five internal/external call treatments to a pager installed in the paging rack

Pager Display

With the existing Radio Paging functionality, when Calling Line Identification (CLID) information is sent to a paging device, a maximum of seven digits are displayed on the pager.

With the Radio Paging Product Improvement Continuation, however, up to 16 digits can be displayed on a pager. Therefore, it is possible for the entire CLID information to be displayed. In order to specify the number of digits (0-16) to be sent to the Radio Paging System, the Transmit Caller's DN (TRDN) prompt must be defined in Overlay 58.

Pretranslation

Pretranslation allows the creation of a flexible dialing plan by using Speed Call lists as Pretranslation Tables. With the Radio Paging Product Improvement Continuation, Pretranslation is activated/deactivated for Radio Paging calls by defining the Pretranslation (PRET) prompt in Overlay 58. This activation/deactivation takes place regardless of whether or not Pretranslation is allowed at a customer level.

Pagers installed in the paging rack

With existing Radio Paging functionality, the treatment of external calls forwarded to pagers in the paging rack is defined by the Recall if busy from Radio Paging (RCAL) prompt in Overlay 58. If RCAL is set to NO, the caller receives a busy tone. If RCAL is set to YES, the call is routed to the attendant. When an internal call is forwarded to a pager in the paging rack, the caller receives a busy tone.

With this Product Improvement Continuation, the user chooses what happens to internal/external calls forwarded to a pager in the paging rack. The treatment of these calls is defined by the Treatment for Internal Calls (INTR) and Treatment for External Calls (EXTR) prompts in Overlay 58.The INTR and EXTR prompts replace the RCAL prompt.

CAUTION

With conversion to X11 Release 23 software, the treatment for external calls to a pager in the paging rack is **not** converted automatically. Therefore, the EXTR prompt must be defined. If EXTR is not defined, when an external call is forwarded to a pager in the paging rack, the call receives the default treatment for external calls (busy tone).

The Radio Paging Product Improvement Continuation offers the following five possibilities for the treatment of calls to pagers in the paging rack:

— The caller receives a busy tone.

— The call is routed to an attendant.

— The caller receives a special tone (SRC1-SRC8) or an announcement (with RAN equipment) delivered from the Tone and Digit Switch (TDS) card.

— The caller receives an announcement from a RAN machine.

— The call is routed to Meridian Mail.

Busy Tone

When INTR or EXTR is set to BUSY, the caller receives a busy tone.

Routed to an Attendant

When INTR or EXTR is set to ATT, the call is routed to an attendant.

Special Tone or Announcement

When INTR or EXTR is set to SRC1-SRC8, the caller receives a special tone, programmed in Overlay 56, or an announcement. After an announcement is provided to the caller, the call is disconnected. Recorded Announcement (RAN) equipment is required to provide this announcement.

Announcement from RAN

When INTR or EXTR is set to RAN, the caller receives an announcement from a RAN machine and is then disconnected or routed to an attendant after the message is heard. Post RAN treatment is defined by the RAN post announcement treatment (POST) prompt in Overlay 16.

For this enhancement to function, a RAN route must be specified by defining the Route number that provides the Recorded Announcement (RANR) prompt in Overlay 58. The RAN route must be specified prior to defining the RANR prompt.

Meridian Mail

When INTR or EXTR is set to MAIL, the call is routed to Meridian Mail. In this case, the caller receives an announcement stating that the call is being rerouted to Meridian Mail. With this enhancement, all Meridian Mail functions are available.

For this enhancement to function, the Meridian Mail Directory Number (MMDN) prompt must be defined in Overlay 58. Prior to defining the MMDN prompt, however, the Voice Automatic Call Distribution (ACD) messages queue must be defined in Overlay 23. The maximum input for Voice ACD is four digits or seven digits if the Directory Number Expansion (Seven Digit) (DNXP) package 150 is equipped.

Operating parameters

The Radio Paging Product Improvement Continuation is applicable on a stand-alone Meridian 1 switch with a Radio Paging system or in an Integrated Services Digital Network (ISDN) Meridian Customer Defined Network (MCDN) with a centralized Radio Paging System.

A maximum of 16 digits can be sent to Radio Paging equipment, as only 16 digits can be stored in the Calling Line Identification (CLID) field.

As per existing Radio Paging functionality, if the calling number is not available, the Route Access Code of the incoming trunk is displayed on the Radio Paging device.

If the calling number is shorter than the specified value defined at the TRDN prompt, the missing digits are replaced by zeros on the pager's display. With the existing functionality, a shorter calling number is also displayed on a pager in this manner.

If the calling number is greater than the specified value defined at the TRDN prompt, the most significant digits are displayed. The unnecessary digits are deleted.

The treatment of calls to a pager in the paging rack is only applicable if the Radio Paging device conforms to the standards of the European Selective Paging Manufacturer's Association (ESPA).

When the Recorded Paging Announcement (PANN) prompt is set to YES in Overlay 58, each redirected call to the paging equipment receives a recorded announcement stating that the called party is being paged. This announcement is provided even if the pager is in the paging rack.

When a pager is in the paging rack and PANN is set to YES, the caller receives an announcement stating that the pager is in the paging rack. After this announcement, the treatment, as a result of the INTR and EXTR prompts, is performed.

When the INTR or EXTR prompts are set to RAN and all Recorded Announcement (RAN) trunks are busy, the caller receives normal ringback tone. As soon as a RAN trunk becomes available, the caller hears a recorded announcement. This is as per existing RAN functionality.

Meridian Mail must be located on the same node as the paging device, in order for calls to a pager in the paging rack to be re-routed to Meridian Mail. If Meridian Mail and the paging device are not located on the same node, an error message appears at the overlay level.

When INTR or EXTR is set to Mail and the maximum number of calls to the Meridian Mail DN exceeds the limit that was set at the MAXP prompt in Overlay 23, the caller receives normal ringback tone. As soon as the number of calls is less than or equal to the MAXP value, the caller receives the recorded announcement or the defined Meridian Mail function. This is as per existing Meridian Mail functionality.

Feature interactions

Radio Paging Product Improvement Continuation has no specific interactions with existing features.

Feature packaging

The Radio Paging Product Improvement Continuation requires the following packages:

— Radio Paging (RPA) package 187, which requires the following package to access Radio Paging:

- Flexible Feature Codes (FFC) package 139

— Pretranslation (PXLT) package 92

The following packages are required for Meridian Mail:

— Make Set Busy (MSB) package 17

— Basic Automatic Call Distribution (BACD) package 40

— Automatic Call Distribution Package A (ACDA) package 45

— Command Status Link (CSL) package 77

— Command and Status Link with Alpha Signaling (CSLA) package 85

— Integrated Message System (IMS) package 35

— Message Waiting Center (MWC) package 46

— End-to-End Signaling (EES) package 10

— Directory Number Expansion (Seven Digit) (DNXP) package 150 for the Meridian Mail DN (MMDN) to contain a maximum of seven digits

The following package is required for Recorded Announcement (RAN):

— Recorded Announcement (RAN) package 7

Feature implementation

Note: The Radio Paging feature must be configured prior to implementing Radio Paging Product Improvement Continuation. If Pretranslation is to be allowed, the Pretranslation feature must also be configured. Depending upon how the INTR and EXTR prompts are defined, Mail and Recorded Announcement (RAN) must be implemented.

LD 58 – Allow or deny Pretranslation.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RPCD	Radio Paging Customer Data Block.
CUST	xx	Customer number.
...		

TRAN	 (TAB) TWO THR FOR NO	Translation type. Translation lookup table (default) Last two digits of DN Last three digits of DN Last four digits of DN No translation (DN sent as PSA code) The TRAN prompt is not given if MRPS = YES. TRAN is then forced to TAB.
- DNLN	0-(4)-16	DN length.
...		
RCTI	0-(30)-120	Time to wait for a "BUSY" transferring set to become idle. After this time, the call is routed to the attendant.
PRET	(YES) NO	Pretranslation for RPA calls (allowed) or denied.

LD 58 – Set the internal and external treatment for calls to a pager in the paging rack, and set the number of digits of the caller's set transmitted to the paging equipment.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RPAX	Radio Paging Access Code Data Block.
CUST	xx	Customer number.
RPAX	nnnn	Radio Paging Access Code. This prompt is repeated to allow multiple entries. Access Codes must be previously defined in LD 57.
ROUT	0-511 0-127	Route number. For Option 11C.
PANN	(NO) YES	Recorded Paging Announcement (denied) or allowed) for this route.
- RPAR	0-511 0-127	Route number that provides the Recorded Announcement. For Option 11C.

INTR	xxxx	Treatment for internal calls to a pager that is in the paging rack.
	(BUSY)	Caller receives a busy tone (default).
	ATT	Call is routed to the attendant.
	SRC1-SRC8	Tones or announcement delivered from the TDS card which is programmed in LD 56.
	RAN	Call is routed to the RAN machine.
	MAIL	Call is routed to Meridian Mail.
- RANR	0-511	Route number that provides the recorded announcement.
	0-127	For Option 11C.
		RANR is prompted if INTR = RAN.
- MMDN	xxxx	Meridian Mail DN which provides the recorded announcement or the defined function. MMDN is prompted if INTR = MAIL. The MMDN may be up to four digits. However, if Directory Number Expansion (DNXP) package 150 is equipped, seven digits are allowed.
EXTR	xxxx	Treatment for external calls to a pager that is in the paging rack.
	(BUSY)	Caller receives a busy tone (default).
	ATT	Call is routed to the attendant.
	SRC1-SRC8	Tones or announcement delivered from the TDS card, programmed in LD 56.
	RAN	Call is routed to the RAN machine.
	MAIL	Call is routed to Meridian Mail.
- RANR	0-511	Route number that provides the recorded announcement.
	0-127	For Option 11C.
		RANR is prompted if EXTR = RAN.
- MMDN	xxxx	Meridian Mail DN which provides the recorded announcement or the defined function. MMDN is prompted if EXTR = MAIL. The MMDN may be up to four digits. However, if Directory Number Expansion (DNXP) package 150 is equipped, seven digits are allowed.
...		
OPER	(AUTO)	Automatic operation (default).
	MANU	Manual operation.
- EXTM	(0)-9	External mode digit for this RPAX. EXTM is prompted when OPER = AUTO.

- INTM	(0)-9	Internal mode digit for this RPAX. INTM is prompted when OPER = AUTO.
- TRDN	(0)-16	Transmit the last x digits of the caller' s DN to the paging equipment. TRDN is prompted if OPER = AUTO.
...		

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	15.59F
Networking:	No

Recall after Parking

This enhancement to the Call Park feature causes a parked call to be recalled to the attendant or night DN if the attendant is in Night Service, rather than to the parking set, if not answered within a customer-defined period of time (two-minute maximum). The call may be external or internal.

Operating parameters

This enhancement does not apply to calls parked by Automatic Call Distribution (ACD) agents.

This enhancement operates in a standalone, but not a in network environment.

Feature interactions

Call Park

This enhancement to Call Park causes a parked call to be recalled to the attendant or night DN if the attendant is in Night Service, rather than to the parking telephone, if not answered within a customer-defined period of time (two-minute maximum). The call may be external or internal.

The recall to the attendant appears on the Recall ICI key. If the attendant is in Night Service, the recall occurs to the night DN. If the night DN is busy, the call is queued if it is an external call.

Feature packaging

The Recall After Parking feature is contained in Call Park (CPRK) package 33.

Feature implementation

LD 50 – Configure Recall after Parking.

Prompt	Response	Description
...		
CPTM	30-(45)-240	Call Park Timer (in seconds). The amount of time a call is held in the parked state before recalling the parking set or the attendant.
RECA	(NO) YES	Recall Attendant. YES = unanswered parked calls recall the attendant. NO = unanswered park calls recall the parking set.

Feature operation

The recall to the attendant appears on the Recall ICI key. If the attendant is in Night Service, the recall occurs to the Night DN. If the Night DN is busy, the external calls are queued.

Recall to Same Attendant

The Recall to Same Attendant (RTSA) feature allows a recall to return to the attendant which last extended the call. If that attendant is busy, the recall is routed to either the first available idle attendant (option RSAA), or queued to the requested attendant until the attendant becomes idle (option RSAX). A call queued to an attendant in this way takes precedence over all other calls. Queued recalls are presented in the order in which they were queued.

The types of calls and recalls which can be queued are as follows:

— inter-attendant calls

— meter recalls

— slow answer recalls

— park recalls

— Camp-on recalls, and

— Call Waiting recalls.

Operating parameters

Attendant recalls brought about by switchhook flash, dial 0, call transfer, conference or the use of a recall key on an SL-1 telephone will not be affected by the RTSA feature.

RTSA will not apply to calls extended by Automatic Call Distribution (ACD) agents.

RTSA is compatible with QCW3 Attendant Consoles, so long as the parameter ALPD is set to NO in LD 12.

If an Attendant Console is maintenance or position busy, then recalls to it will be presented to the first idle Attendant Console, no matter which option has been specified.

If an attendant fails to answer a direct recall, that Attendant Console is forced into position busy, and the recall is presented to the first idle attendant.

RTSA is not supported by Centralized Attendant Service (CAS).

If the customer enters Night Service while recalls are timing for RTSA, these recalls will not be directed to the night station.

Feature interactions

AC15 Recall: Timed Reminder Recall

With the AC15 Timed Reminder Recall feature, if RTSA = RSAA the call is presented to the attendant who last extended the call, if RTSA = RSAX the call is presented to the attendant who last extended the call or put in the queue if this attendant is busy.

Attendant Forward No Answer

If the attendant does not answer a call and the Attendant Forward No Answer feature is equipped, the console is forced into the Position Busy state and the call routed to the first available idle attendant.

Attendant Overflow Position

Recalls and inter-attendant calls are not routed to the Attendant Overflow Position.

Attendant Position Busy

If an Attendant Console is in maintenance or Position Busy when a Recall to Same Attendant call is recalled to it, the recall is presented to the first available idle attendant. If an attendant goes into Position Busy with a Return to Same Attendant call in Call Waiting, the waiting call is presented to the first available attendant.

Automatic Call Distribution

Recall to Same Attendant does not apply to calls extended by Automatic Call Distribution agents.

Call Forward No Answer

If the attendant does not answer a call and the Attendant Forward No Answer feature is equipped, the console is forced into the Position Busy state and the call routed to the first available idle attendant.

Call Waiting Options

All options for call-waiting calls do not apply to calls queued to a specified attendant. The exception to this is the display call waiting key, which shows the number of calls in the overall attendant queue and the calls in the queue for a specified attendant.

Centralized Attendant Service

Centralized Attendant Service does not support the Recall to Same Attendant feature.

Flexible Attendant Call Waiting Thresholds

The Recall to Same Attendant (RTSA) feature has precedence over the Flexible Attendant Call Waiting Thresholds (FACWT) feature. If either RSAA or RSXA options are selected, RTSA has precedence over FACWT in determining the Call Waiting Lamp state. If one or more RTSA calls are waiting in the attendant queue, RTSA will set the Call Waiting Lamp state to wink (30 impulses per minute).

RTSA calls are not included when the FACWT feature determines the number of calls waiting.

Group Hunt

Calls redirected from a group hunt list via the listed DN or flexible attendant DN, and transferred back to the Pilot DN, are recalled if the Slow Answer Recall Timer expires. However, in practical configurations, the hunt terminates on the entry with the listed DN or attendant DN before the Slow Answer Recall Timer expires; consequently, the call is not redirected to that DN and presented on the applicable ICI key on the console. Therefore, the call is never presented as a recall, so that Recall to the Same Attendant does not apply.

Idle Extension Notification

An Idle Extension Notification recall will always recall to the same attendant, regardless of the configuration of the Recall to Same Attendant (RTSA) feature.

Multi-Party Operations

Users of analog (500/2500 type) telephones can perform an attendant recall during a two-party connection by performing a switchhook flash and then dialing the attendant DN.

Multi-Tenant Service

If a specified attendant is in maintenance or Position Busy, the recall first tries to terminate at another attendant within the same console group, and then to the night DN.

Network Attendant Service

This feature operates on a network-wide basis for the following call types:

— Slow Answer Recall

— Camp-on Recall, and

— Call Waiting Recall.

The operation of this feature is affected by the programming for the option in the Customer Data Block of the system where the attendant answering the call resides.

Periodic Pulse Metering

Meter recalls are returned to the same attendant whether Recall to Same Attendant is allowed or not. If Return to Same Attendant with Queuing on Busy (RSAQ) is selected as an option, the recalls are queued to a specified attendant.

Ring Again on No Answer

A telephone that is recalling the attendant cannot apply Ring Again on No Answer.

Voice Messaging

Recall to Same Attendant does not apply to recalls from the Voice Messaging System.

Feature packaging

Recall to Same Attendant is included in base X11 system software.

Feature implementation

LD 15 – Modify data for each customer member to be configured.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add, or change.
TYPE	CDB ATT	Customer Data Block. Release 21 gate opener.
...		
RTSA	(RSAD) RSAA	Recall to same attendant (denied) allowed.
	RSAX	Recall to same attendant allowed, with queuing on busy attendant.

Feature operation

If the requested attendant is idle, a recall to it will be presented on the loop key, and on the corresponding MTR, IAT, or RLL Incoming Call Indicator (ICI) key.

When a recall is queued specifically for an attendant, this will be indicated on the Attendant Console by a wink lamp state for the Call Waiting lamp.

Introduced in X11 Release:	15.59F
Networking:	No

Recall with Priority during Night Service

This feature (RPNS) places a priority level on the order in which calls queued to a Night DN are processed as follows:

— recall of an external call

— a new external call, and

— other calls.

This is the normal order during day processing.

Operating parameters

Due to the prioritizing of call processing, low priority calls may remain queued for a long time before being processed.

Feature interactions

None

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 15 – Configure Recall with Priority during Night Service.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	CDB NIT	Customer Data Block. Release 21 gate opener.
...		
- RPNS	(NO) YES	(Deny) allow Recall with Priority during Night Service.

Feature operation

The recall to the attendant appears on the Recall ICI key. If the attendant is in Night Service, the recall occurs to the Night DN. If the Night DN is busy, the external calls are queued.

If there is an occurrence of several calls of the same type to a station, the calls are presented to the station in their chronological order of arrival.

Introduced in X11 Release:	All
Networking:	No

Recorded Announcement

The Recorded Announcement (RAN) feature allows the Meridian 1 to connect calls automatically to a customer-provided Recorded Announcement machine. Recorded Announcements can be used for:

— Automatic Call Distribution (ACD)

— Automatic Wake Up

— Intercept Treatment (INTR)

— Recorded Overflow Announcements (ROAs), and

— Network Queuing feature, which has Call Back Queuing (CBQ), Coordinated Call Back Queuing (CCBQ), Call Back Queuing to Conventional Main (CBQCM), and Off-Hook Queuing (OHQ).

The system software detects calls to connect to the Recorded Announcement (RAN) machine, determines the Intercept Treatment required, and connects the call to the proper Recorded Announcement. The system then monitors the RAN machine.

The Meridian 1 provides the software programs to control the announcement recorder and the circuit packs. Two types of circuit packs can be used:

— Recorded Announcement (RAN) Trunk Cards (QPC74) contain four identical trunk circuits for the interface between the Meridian 1 and the announcement machine. See *QPC74 Recorded Announcement Trunk Card description* (553-2201-194) for engineering information. When the QPC74 is used, all ports on the card must be dedicated as TYPE RAN or TYPE MUS.

— Universal Trunk Cards (NT8D14AA) contain eight identical trunk circuits that can be configured independently in the system software. See *NT8D14 Universal Trunk Card description* (553-3001-171) for a description.

Operating parameters

Dial access to RAN trunk groups is allowed and is limited only by Trunk Group Access Restrictions (TGARs).

When the QPC74 is used, all ports on the card must be dedicated as TYPE RAN or TYPE MUS.

Feature interactions

Conference
No Hold Conference

A RAN trunk cannot be Conferenced or No Hold Conferenced.

Collect Call Blocking

A RAN route is defined as having CCBA YES or NO, which is used if Coordinated Dialing Plan (CDP) or ACD queues were not used to get to the RAN route. If the call is routed through ACD/CDP to terminate on RAN, the Collect Call Blocking (CCB) treatment will depend upon the CCB data of the ACD/CDP, and not of the RAN route.

FCC Compliance for DID Answer Supervision

With FCC Compliance for DID Answer Supervision, incoming DID calls that are intercepted to a Recorded Announcement (RAN) are provided with answer supervision.

Group Hunt

Calls which are queued against the Group Hunt Pilot DN cannot receive Recorded Announcement.

Recovery on Misoperation of Attendant Console

If a Recorded Announcement is given to the destination side that has been intercepted, the connection to the destination side is considered as invalid. Therefore, if the attendant tries to extend the source to the destination using the RELEASE key or another LOOP key, the operation is ignored. The attendant must first press the RELEASE DESTINATION key to release the destination, and then extend the call to the source. If the HOLD key is pressed, the source party is put on hold and the Recorded Announcement is disconnected on the destination side.

Source Included when Attendant Dials

The source is included in a conference involving the attendant, the source, and Recorded Announcement or music treatment. Intrusion tone is not provided in this case.

Trunk Traffic Reporting Enhancement

The Trunk Seizure Option is not supported on RAN trunks.

Feature packaging

Recorded Announcement (RAN) package 7, which requires Intercept Treatment (INTR) package 11.

Feature implementation

LD 16 – Add or change Recorded Announcement (RAN) trunk route.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
CUST	0-99 0-31	Customer number, as defined in LD 15. For Option 11C.
ROUT	0-511 0-127	Route number. For Option 11C.
TKTP	RAN	RAN trunks.

RTYP	CAP	Code-a-Phone recording device. Software allows announcements of up to 320 seconds in length in X11 Release 14, or 608 seconds in X11 Release 15.
	AUD	Audichron recording device (required when connecting to a Universal Trunk Card). Software allows announcements of up to 64 seconds.
	CK2	Cook Electric recording device. Software allows announcements of up to 64 seconds.
	DGT	Digital Recorders 213300 & 213400. Software allows announcements of up to 256 seconds on X11 Release 15 and later.
	CON	NT7M series digital recorders. Software allows announcements of up to 608 seconds on X11 Release 15 and later.
REP	1-15	Number of times the announcement repeats during each connection.
POST	ATT	Call is routed to attendant after specified number of repetitions (applies to Direct Inward Dial [DID] calls on Intercept).
	DIS	RAN is removed after a specified number of repetitions.
STRT	IMM	Call connects immediately to announcement.
	DDL	Call connects to announcement at the start of announcement.
ASUP	(NO) YES	Supervision (is not) or is required to inform the Central Office (CO) when the call is answered.
ACOD	xxx...x	Trunk route access code.

Note: All RAN route members must be removed before the route can be removed.

LD 14 – Add or change Recorded Announcement (RAN) trunk.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RAN	RAN trunk data block.
TN	l s c u c u	Terminal Number. For Option 11C.
CUST	0-99 0-31	Customer Number, as defined in LD 15 (prompted if REQ = NEW). For Option 11C.
RTMB	0-511 0-510 0-127 0-510	Route and member number. For Option 11C.

Note: If a night table is used with Network Automatic Call Distribution (NACD), the FROA and FRT values in LD 23 need to be set for the Recorded Announcement feature. FROA should be "NO" and FRT should be four seconds greater than the last entry time of the night table.

Feature operation

No specific operating procedures are required to use this feature.

Recorded Announcement Broadcast

The Recorded Announcement Broadcast (RANBRD) feature expands the existing functionality of the Recorded Announcement (RAN) feature. Previously, the Recorded Announcement (RAN) feature used one-to-one connection between a calling party and a designated RAN trunk connected to a physical Recorded Announcement machine. Therefore, if four calling parties were receiving RAN treatment then four RAN trunks were occupied to provide this functionality.

The Recorded Announcement Broadcast feature eliminates the need for multiple cross-connections to provide recorded announcement. With this feature, multiple calling parties receive RAN treatment from one RAN trunk. Thus allowing a RAN trunk to simultaneously broadcast announcements to maximum of 48 calling parties per RAN trunk. This expansion maximizes the usage of available RAN trunks.

This feature also introduces the following enhancements:

— Incremental Software Management limits

— RAN signalling capabilities

— Multi-Channel RAN Machine Types and Modes

— Message Staging Through Queuing Thresholds for Delay Dial Start/Stop RAN machines

— Music on Waiting

— Traffic Study Option

Each of the above enhancements are discussed in the sections that follow.

Incremental Software Management Limits

Two new Incremental Software Management (ISM) limits on Broadcast Routes and Broadcast Connections are introduced with this feature.

Overlay 22 is modified to print the new ISM information on RAN Broadcast connections that is introduced for the RAN Broadcast feature. The existing SLT command prints the ISM information for the system.

Option 11C and Input-Output Disk Unit with CD-ROM (IODU/C) customers can modify ISM parameters via keycode. A keycode is a machine-generated digitally signed list of customer capabilities and authorized software release. A security keycode scheme protects ISM parameters.

In order for Option 11C and IODU/C customers to expand ISM limits, they must order and install a new keycode. This installation is performed using the Keycode Management feature. All Keycode Management commands are executed in Overlay 143. To make the expansion effective, the customer must sysload. For further information on keycode installation, please refer to *Software conversion procedures.*

For customers without Option 11C or IODU/C, ISM parameters are delivered as per existing operation.

For further information on ISM, refer to the Incremental Software Management feature module in *X11 features and services*.

Broadcast Routes

The ISM limit on broadcast routes is based on the number of broadcasting RAN routes available on a system. A new ISM header in Overlay 16 indicates ISM broadcasting RAN information for the system. This information is updated as each new RAN broadcasting route is configured by the customer. The upper ISM limit for broadcast routes is 511 for Options 51C-81C and 127 for Options 11C. Table 133 shows the Broadcast RAN Route ISM information that is added to the header in Overlay 16.

Table 133
New Broadcast RAN Routes ISM Information in Overlay 16

RAN RTE	AVAIL: xx	USED: xx	TOT: xx

Broadcast Connections

The ISM limit on broadcast connections is based on the number of broadcast RAN connections available on the system. Additional broadcast RAN connections can be purchased incrementally. A new ISM header in Overlay 14 indicates ISM broadcasting RAN connections ISM information for the system. Table 134 shows the Broadcast RAN Connections ISM information that is added to the header in Overlay 14.

Table 134
New Broadcast RAN Connections ISM information in Overlay 14

TNS	AVAIL: xxxxx	USED: xxx	TOT: xxxxx
RAN CON	AVAIL: xxxx	USED: xxx	TOT: xxxx

As each new broadcasting RAN trunk is configured, the number of available broadcast connections is subtracted from the maximum number of broadcast connections to the RAN trunk. Any calling party that is listening to a recorded announcement through a broadcasting RAN trunk represents a broadcast connection.

The following scenario provides a detailed example of the new ISM limits that are applicable to this feature. Assume that a customer has an upper ISM limit of 5 broadcast RAN routes and an upper ISM limit of 240 broadcast connections. When the customer defines a new broadcast RAN route, the new number of available broadcast RAN is equal to the upper limit less 1, in this case that would be 4 broadcast RAN routes. When the customer configures 2 RAN trunks for the RAN route in Overlay 14 and 16 broadcast connections to each trunk. The number of available broadcast connections is now equal to the upper limit less the number of configured broadcast RAN connections. So, in this scenario the customer has a total of 208 (240 -16-16=208) broadcast connections and a total of 4 broadcast RAN routes.

RAN Signaling

Immediate Start

With immediate start RAN signaling, the calling party is connected to the recorded announcement immediately. With this signaling, calling parties barge-in on the announcement. Therefore, the calling party can be connected to the announcement such as the beginning, middle or end.

The RAN Broadcast feature allows immediate start configuration the option of receiving Music On Hold to calling parties waiting for RAN treatment.

Delay Dial

With delay dial RAN signaling, the calling party is only connected at the start of a recorded announcement. With RAN Broadcast, calling parties can have the option of Music On Hold while waiting for the start of the announcement.

Multi-Channel RAN Machine Types and Modes

Multi-Channel corresponds to multiple RAN channels that can be configured within one RAN trunk route. In a Multi-Channel RAN route, each trunk has its own dedicated RAN channel on a physical RAN machine. Multi-Channel RAN routes do not support the cross connecting (daisy chains) of multiple trunk ports together so that several callers hear the same RAN message.

As an example in Multi-Channel RAN configuration, a Level Start/Stop Multi-Channel (MLVL) route could have trunk ports each configured with it's own RAN channel. Each trunk could be assigned several RAN Broadcast connections. If the message is 15 seconds long, then queuing could be configured to start playing a message every 3 seconds.

The new multi-channel machine types - Continuous Mode Multi-Channel (MCON), Pulse Start/Stop Multi-Channel (MPUL) and Level Start/Stop Multi-Channel (MLVL) - are not linked to RAN machine or a given trunk. All trunks belonging to the RAN route are considered independent. RAN trunks and RAN machine channels are connected one to one. Accordingly, if one RAN trunk is detected as faulty then all other trunks are not impacted.

For these new RAN machine types, the maximum length of the recorded announcement is configured is two hours. The meaning of a ground signal received from the RAN machine (play or idle) is configured in Overlay 16. This prompt was previously only applicable to XFEM RAN trunks.

These new RAN machine types are applicable to broadcasting and non broadcasting RAN routes.

Recorded Announcement Broadcast supports two machine modes: Continuous and Start/Stop. Both modes support immediate start and delay dial configurations.

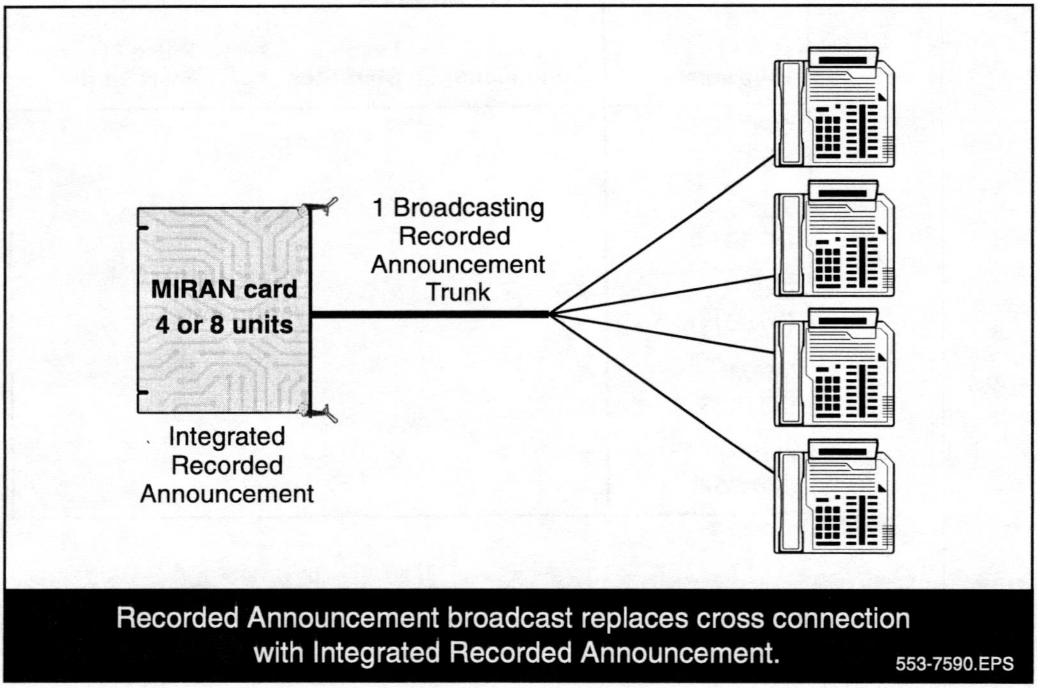
Table 135 outlines the hardware requirements and new RAN modes. RAN Broadcast requires an external RAN machine and a RAN trunk card.

Table 135
RAN modes and Hardware

Hardware	Types of RAN Modes: Continuous	Level Start/Stop	Pulse Start/Stop
QPC (X74)	X		X
XUT (NT8D14)	X		
EXUT (NT8D14)	X	X	X
XFEM (NT5K83)	X		X
MIRAN (NTAG36)	X	X	

As shown in Figure 74, the Meridian Integrated Record Announcement (MIRAN) cards eliminates the need for an external RAN machine. MIRAN emulates the Extended Universal Trunk (EXUT) card capabilities and provides built-in, physical RAN channels.

Figure 74
MIRAN Hardware



Continuous Mode

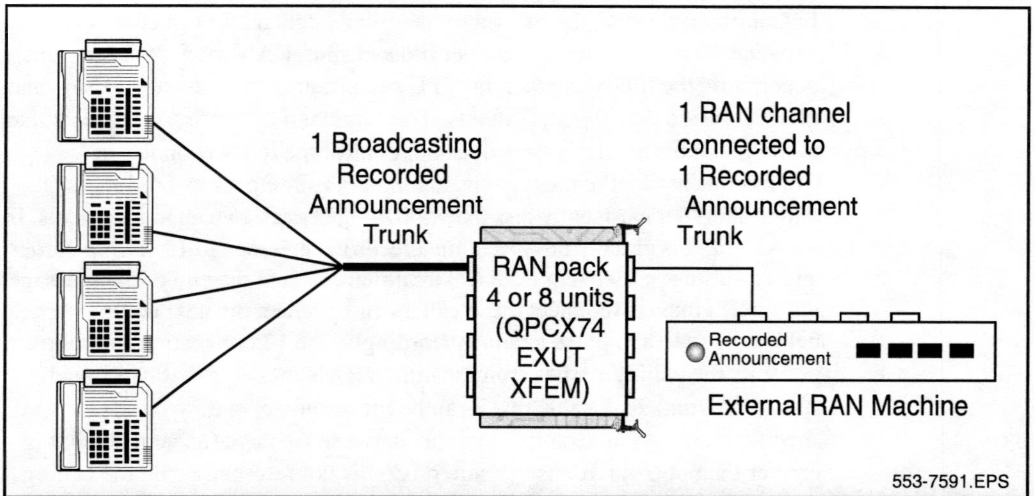
In Continuous mode, the recorded message is repeatedly played over and over. Calling parties requiring RAN treatment barge in on a playing message or receive ringback tone until the message starts over.

In Continuous mode, the maximum recommended amount of connections is between 10 and 16 connections per broadcasting RAN trunk. This amount depends on the following factors: CPU performance, answer supervision and delay between two announcements. This engineering requirement exists due to the fact that the Meridian 1 does not control the RAN channel. In Continuous mode, the message is continually running a recorded message with a short delay (usually less than 500 ms) between two announcements. If a RAN trunk is already broadcasting a recorded message to 12 calling parties and 12 calling parties require RAN treatment, then at the end of the message the system must disconnect these callers and connect the next calling parties before the message plays again. Accordingly, the 12 connection limitation prevents the calling parties from hearing a RAN message that has already started playing. This value of 12 can be increased depending on the system specifications. As an example, a delay between two announcements that is greater than 500 ms. If answer supervision is not returned when the calling party connects to the recorded announcement, then up to 24 connections per Continuous mode RAN trunk are supported.

Recorded Announcement Broadcast introduces a new Continuous mode machine type called Continuous Mode Multi-Channel (MCON). Independent (asynchronous) RAN trunks can belong to a MCON RAN route which was not permitted with the existing Audichron/Cook 211 (AUD), NT7M Digital Recorders (CON) or 213300 and 213400 Digital Recorders (DGT) Continuous mode machine types.

Figure 75 illustrates RAN Broadcast using Start/Stop or Continuous Mode configuration.

Figure 75
RAN Broadcast using Start/Stop or Continuous Mode Configuration



With 1 Recorded Announcement broadcast, 1 single Recorded Announcement Channel may broadcast up to 48 calling parties using one single RAN trunk.

Start/Stop Mode

In Start/Stop mode, the recorded message does not begin to play the recorded announcement until a start pulse signal is received from the RAN machine. There are two types of Start/Stop mode: Pulse Start/Stop and Level Start/Stop.

With Start/Stop configuration, the system controls when the RAN message starts and stops. Therefore, if 30 calling parties require RAN treatment, the Meridian 1 waits to start the recorded announcement until all 30 callers are ready to be connected. When the message is finished playing, the system disconnects all 30 callers and waits until the next 30 callers are queued before sending a message to the RAN machine to start playing the message. The recommended value for the maximum number of connections per broadcast start/stop trunk is 30. Again, this value can be increased depending on the system specifications. As an example, if the answer supervision signal is not returned when the calling party connects to the recorded announcement, then up to 48 connections per Start/Stop RAN trunk are supported.

With Pulse Start/Stop, the start signal is pulse. This pulse activates the playback of the recorded announcement. The announcement is played until completion. All other start pulses are ignored until the announcement has finished.

With Level Start/Stop, the start signal is a level. The leading edge of the start signal initiates the playback of the recorded announcement. This continues until either the trailing edge of the start signal occurs or the announcement has finished. When a trailing edge is detected, the recorded announcement is terminated and level start signal is sent to the RAN machine to immediately reset the recorded announcement.

Recorded Announcement Broadcast introduces two Start/Stop mode machines types called Pulse Start/Stop Multi-Channel (MPUL) and Level Start/Stop Multi-Channel (MLVL).

Message Staging

Recorded Announcement Broadcast allows the staging of recorded announcement for Delay Dial Start/Stop Machines. The staging of announcements is controlled by the queuing thresholds programmed in Overlay 16 for Delay Dial Start/Stop machines. With staging, if several copies of a recorded announcement are available on different RAN ports, then the start time of the recording can be staggered. For queued calling parties, this decreases the waiting time to hear the start of the announcement.

In Continuous modes, the staging of announcements is determined by the RAN machine.

Queuing Thresholds for Delay Dial Start/Stop Machines

The Recorded Announcement Broadcast feature introduces two new queuing thresholds for Start/Stop RAN machines configured with Delay Dial signaling (STRT=DDL in LD 16).

These new queuing thresholds allow customers to stagger recorded announcements using both time and number of calls as threshold triggers. Queuing thresholds optimizes a calling party's waiting time and the number of calls waiting to receive RAN treatment.

As an example, a customer has a recorded announcement that is 15 seconds in length. This announcement is used in a high volume Automatic Call Distribution (ACD) environment. In this scenario, a calling party requiring RAN treatment can range between 1 to 30 at any given time. With RAN Broadcast the 15 second message can be staggered. With this arrangement, 5 trunk ports could be configured in a RAN broadcast route with each trunk provisioned with 10 RAN broadcast connections. The message could then programmed to play every 3 seconds or when 10 caller are queued (TITH = 3 and NCTH = 10 in LD 16). In this configuration, each of the 5 trunks would be connected to individual RAN channels with each channel having the identical 15 second message. The calling party would only have to wait a maximum of 3 seconds before receiving a recorded announcement message.

With the new queuing thresholds, when the waiting or the number of calls threshold is met or exceeded the system searches for an available RAN Trunk and connects all queued callers waiting for a recorded announcement. If the system cannot locate an available trunk, then the waiting calls are requeued without a threshold so that waiting callers are connected to a RAN trunk as soon as it becomes available.

However, if RAN trunks are not available then callers are requeued without a threshold until the next RAN trunk is available. At this point, all threshold exceeded callers listen to the recorded announcement.

If no time or number threshold is configured, then all queued parties are connected to the first available RAN trunk. This includes callers that have just been queued by the system. Therefore, the system does not assign any priority to waiting callers when no thresholds have been configured.

Music on Waiting

Recorded Announcement Broadcast feature supports music on waiting for queued callers on both broadcasting and non-broadcasting RAN trunks. With this enhancement, music is provided when a calling party is queued to receive a recorded announcement. A selected music source is provided to waiting callers until the system locates an available RAN trunk. The music on waiting enhancement replaces ringback tone.

Traffic Study Option

The Traffic Period Option (TPO) allows a customer to enhance their TFC002 reports to accumulate trunk usage data after every traffic period instead of accumulating usage only after a call disconnects. With this option enabled in Overlay 17, the Common Channel Signaling (CCS) associated with lengthy calls is reported in each traffic report interval throughout the duration of the call.

Previously, this feature did not apply to RAN and Music trunks. However, with the introduction of the RAN Broadcast feature, changes are made to the Trunk Traffic Reporting Enhancement with the introduction of TFC111. The TFC111 report provides information on the usage of broadcasting routes. For the TFC111 to be output, the customer report number **11** must be selected using the SOPC command in Overlay 2. For example, for Customer 0, SOPC 0 11 is entered. To print the TFC111 report, the TOPC command in Overlay 2 is used. For example, for Customer 0, TOPC 0 11 is entered. The TFC 111 report is also printed when automatic traffic reports are scheduled in Overlay 2.

A traffic message is output each time the number of active broadcasting connections is equal to the system's ISM limit.

The new TFC111 report provides the following information:

— the trunk type

— the number of successful broadcast connections of the trunk associated with route

— the average duration of broadcast connects for route

— the average waiting time for RAN requests

— the maximum waiting time for RAN requests

— the waiting time threshold peg count

— the number of waiting parties threshold peg count

— the broadcast connection peg count for three lowest usage trunks

Figure 76 is an example of the customer report, TFC 111, for RAN Broadcast routes.

Figure 76
New Customer Traffic Measurement Outputs

System ID 0200	TFC111	
Customer Number 000		
Route Number 031	Trunk Type RAN	
Successful broadcast connections peg count 000817	Average call duration 00006	Average waiting duration 00004
Maximum waiting time 00007	Waiting time threshold peg count 00000	Number of waiting parties threshold peg count 00000
Broadcast connections peg count for lowest trunk usage 00000	Broadcast connections peg count for next to lowest trunk usage 00000	Broadcast connection peg count for second lowest trunk usage 00002

Maximum number of connections per broadcasting RAN trunk

Table 136 shows the maximum number of connections per broadcasting RAN trunk that can be configured. These values depend on system configuration; therefore, some systems can allow greater values or request lower values.

When no answer supervision signal is to be returned at the time the caller receives the announcement, more connections are supported. This is the case with unsupervised trunks, internal calls, or when the answer signal has already been sent.

If answer supervision is returned, there is a high impact on realtime. Therefore, it is recommended that the maximum number of connections per RAN trunk be set to a lower value (See Table 136).

To achieve maximum efficiency, TFC111 and the TITH and NCTH thresholds can be used. For instance, the difference between the number of times TITH was met and NCTH was met provides an indication of how the system reacts to the incoming RAN request rate. In the case of a high rate, a greater number of NCTH was met than TITH. This indicates that the number of connections is insufficient.

Table 136
Recommended maximum number of connections per trunk

RAN mode	Is answer supervision returned when RAN is provided?	Recommended maximum number of connections per RAN trunk
Continuous mode with less than 500ms between two announcements	Yes	up to 12
Continuous mode with less than 500ms between two announcements	No	up to 24
Start/Stop mode	Yes	up to 30
Start/Stop mode	No	up to 48

Operating parameters

The Recorded Announcement Broadcast feature is applicable to RAN routes only.

The Meridian Integrated Recorded Announcement (MIRAN) card provides a multi-tasking environment for certain voice processing intensive applications, such as RAN and Music on Hold. This card stores recorded music and announcements in flash memory using two audio ports. The setup or modification of sound files is done using a set or a TTY. This card stores recorded music and announcements in flash memory or PCMCIA flash memory cards. Music can be played from an analog source, such as a Compact Disc (CD) player or a Muszac source, through the MIRAN card. It is not a requirement that Music be recorded within MIRAN. The card plays music from other sources.

When configuring this feature, the mode supported by the external RAN machine and Meridian 1 hardware must match. The EXUT card supports continuous, pulse start/stop and level start/stop. The XFEM card supports continuous and pulse start/stop modes. The MIRAN card supports continuous and level start/stop modes.

The MIRAN card is associated with a certain port on the EXUT card. Each recorded announcement can be associated with more than one port at one time.

Traditional Recorded Announcement and Recorded Announcement Broadcast can exist on the same system.

If using a Start/Stop RAN machine, it is recommended that both the Waiting Time Threshold (TITH prompt) and the Number of Calls Waiting Threshold (NCTH prompt) be configured.

The Waiting Time Threshold (TITH) and the Number of Calls Waiting Threshold (NCTH) prompts should be configured to minimize caller's waiting time. TITH should be set to the length of the RAN message divided by the number of RAN trunks. NCTH should be set to the maximum number of connections per trunk divided by the number of RAN repetitions. All RAN trunks should have the same number of allowed connections to trigger RAN starts.

The continuous mode multichannel, the level start/stop multichannel and the pulse start/stop multichannel all support independent RAN trunks.

A RAN route can be modified to disallow broadcasting, provided that all trunks do not have any active calls connected when changes are made. When modifying a RAN route to allow broadcasting, the number of available ISM RAN connections must be sufficient or the change is not permitted.

In customer situations with high RAN usage, continuous RAN is recommended. In situations with a fluctuating or low incoming rate, a start/stop RAN with thresholds configured at a low value is recommended.

Feature interactions

Answer Supervision

Answer Supervision is provided based on the configuration of the RAN route. When music is provided to queued callers waiting for an announcement, the answer supervision is returned as though the recorded announcement was given.

Automatic Call Distribution Recorded Overflow Announcement

Automatic Call Distribution (ACD) and Recorded Overflow Announcement (ROA) allows queued calls to an ACD agent or attendant to be routed to a recorded announcement informing the calling party of the delay. If music is selected between the first and second recorded announcement, queued calls can be routed to a second announcement if they are still waiting in the queue.

When Music on Waiting is configured for the second RAN route, the music source selected by the Automatic Call Distribution or Recorded Overflow Announcement feature, already provided to a queued call, is not replaced by the one selected by the second RAN route when this queued call is waiting to be connected to the second RAN.

Automatic Wake Up

Automatic Wake Up (AWU) broadcast capability is independent of the RAN broadcast capability. AWU broadcast is only applicable to AWU trunks.

Incremental Software Management

The Incremental Software Management (ISM) limits introduced by this feature impact the number of units available and used by the ISM feature. The ISM header at the start of Overlay 14 is updated to indicate the broadcast RAN connections ISM information on the system.

INIT ACD Queue Call Restore

ACD calls queued for receiving RAN are restored by the INIT ACD Queue Call Restore feature following system initialization. All other calls queued for RAN are dropped, and the callers hear silence.

If system initialization occurs when an Automatic Call Distribution (ACD) call is being greeted by ACD RAN, the RAN greeting is automatically disconnected. If the call is restored by the INIT ACD Queue Call Restore feature, the call is presented to the appropriate ACD Directory Number as a new call.

When system initialization occurs, Music on Waiting is stopped and the restored call is presented to the ACD DN as a new call.

Integrated Call Center Management

Integrated Call Center Management (ICCM) broadcast capability is independent of the RAN Broadcast capability. ICCM broadcast is only applicable to IVR voice ports.

The script command GIVE RAN<RAN route number> connects a call to the specified RAN route and the RAN broadcast feature will apply if applicable.

Music Broadcast

The Music Broadcast feature is applicable to Music only, and the RAN Broadcast feature is applicable to RAN only.

Feature packaging

The Recorded Announcement Broadcast (RANBRD) feature is package 327. The following packages are also required:

— Recorded Announcement (RAN) package 7

— Intercept Treatment (INTR) package 11

Feature implementation

The following scenario provides details on how to configure RAN Broadcasting and Non Broadcasting using different applications such as Automatic Call Distribution (ACD) queues and intercept treatments.

Assume the following scenario exists. You have a Meridian 1 configured with non-broadcasting RAN. Your system has 3 RAN routes. Route 1 has 1 trunk with low usage and handles RAN intercept treatments. Route 2 has 8 trunks with variable usage and handles Recorded Overflow Announcement (ROA). Route 3 has 16 trunks with high usage and handles all Automatic Call Distribution (ACD) greetings into your call centre.

Tables 137 and 138 provide a non-broadcasting and a broadcasting scenario respectively.

Table 137
Non-Broadcasting Scenario

RAN Routes	Usage	RAN Mode	Number of Trunks	RAN Machine Type
1	Low	Start/Stop	1	Cook 201/QAY1.
2	Varied	Continuous	8	Audichron/Cook 211 (required for XUT trunks)
3	High	Continuous	16	Audichron/Cook 211 (required for XUT trunks)

In the non-broadcasting scenario the following system requirements exist:

— a total of 25 (1 + 8 + 16) RAN trunks

— a total of 3 RAN channels

When using the RAN Broadcast feature in the same scenario, RAN trunks and RAN channels requirements are reduced. With this feature, each group of RAN trunks is replaced by one broadcast RAN trunk with maximum number of connections set to the number of cross connected trunks. RAN Broadcast allows a maximum of 48 connections per RAN trunk.

Table 138
Broadcasting Scenario

RAN Routes	Usage	RAN Mode	Number of Trunks	RAN Machine Type	Broadcast Connection/ Trunk
1	Low	Start/Stop	1	Cook 201/QAY1	non broadcast
2	Varied	Continuous	1	Audichron/Cook 211 (required for XUT trunks)	8 connections
3	High	Continuous	1	Audichron/Cook 211 (required for XUT trunks)	16 connections

In the broadcasting scenario the following system requirements exist:

— a total of 3 (1+1+1) RAN trunks

— a total of 3 RAN channels

— a total of 2 RAN Broadcast Route ISM limits

— a total of 24 (8 + 16) RAN Connections ISM limits

The broadcasting scenario can be further enhanced if RAN routes 2 and 3 used a Delay Dial Start/Stop RAN trunk with the Number of Calls Waiting Threshold and Waiting Time Threshold configured.

LD 16 – Define Continuous RAN Route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-511 0-127	Route Number. For Option 11C.
TKTP	RAN	Recorded Announcement trunk type.
RTYP	 AUD CON DGT MCON	Recording devices for RAN trunks where: Audichron/Cook 211 (required for XUT trunks). NT7M Digital Recorders (X11 Release 15). 213300 and 213400 Digital Recorders (X11 Release 15). Continuous mode Multichannel.
REP	1-15	Number of repetitions of recorded announcements.
POST	 ATT DIS	RAN Post announcement treatment where: Route to attendant after maximum repetitions Disconnect after maximum repetitions.
STRT	 IMM DDL	Start arrangement where: Immediately connect call to recording. Delay call connection until start of recording.
WAIT	RGB	Provide ringback for call queuing for RAN trunk (default). MUS = Provide music for calls queuing for RAN trunk.
- MRT	0-511 0-127	Music route for RAN queuing. For Option 11C. MRT is only prompted for RAN route with WAIT = MUS.
BDCT	YES	Allow RAN broadcast for this route. NO = Deny RAN broadcast for this route (default).

ASUP	(NO)	Do not return answer supervision (default). YES = Return answer supervision. CO = Return answer supervision only if originator is a Central Office trunk.
ACOD	x...x	Access Code for the trunk route. The Access Code must not conflict with the numbering plan. ACOD can be four digits, or seven digits with DNXP package 150 equipped.

LD 16 – Define Immediate Start/Stop RAN Route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-511 0-127	Route Number. For Option 11C.
TKTP	RAN	Recorded Announcement trunk type.
RTYP	 CAP CKM PUL LVL MPUL MLVL	Recording devices for RAN trunks where: Code-A Phone. Cook 201 multichannel. Pulse start/stop (X11 Release 19 Enhanced Universal Trunk cards). Level start/stop (X11 Release 19 Enhanced Universal Trunk cards). Pulse start/stop multichannel. Level start/stop multichannel.
REP	1-15	Repetitions of recorded announcements.
POST	aaa	RAN Post announcement treatment where: ATT = Route to attendant after maximum repetitions DIS = Disconnect after maximum repetitions.
STRT	IMM	Immediately connect call to recording.
WAIT	RGB	Provide ringback for call queuing for RAN trunk (default). MUS = Provide music for calls queuing for RAN trunk.

- MRT	0-511 0-127	Music route for RAN queuing. For Option 11C. MRT is only prompted for RAN route with WAIT = MUS.
BDCT	YES	Allow broadcast capability for this route. NO = Deny broadcast capability for this route (default).
ASUP	(NO)	Do not return answer supervision (default). YES = Return answer supervision. CO = Return answer supervision only if originator is a Central Office trunk.
ACOD	x...x	Access Code for the trunk route. The Access Code must not conflict with the numbering plan. ACOD can be four digits, or seven digits with DNXP package 150 equipped.

LD 16 – Define Delay Dial Start/Stop RAN Route.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-511 0-127	Route Number. For Option 11C.
TKTP	RAN	Recorded Announcement trunk type.
RTYP	 CAP CK2 CKM PUL LVL MPUL MLVL	Recording devices for RAN trunks where: Code-A Phone. Cook 201/QAY1. Cook 201 Multichannel. Pulse start/stop (X11 Release 19 Enhanced Universal Trunk cards). Level start/stop (X11 Release 19 Enhanced Universal Trunk cards). Pulse start/stop multichannel. Level start/stop multichannel.
REP	1-15	Repetitions of recorded announcements.

POST	aaa	RAN Post announcement treatment where: ATT = Route to attendant after maximum repetitions DIS = Disconnect after maximum repetitions.
STRT	DDL	Delay call connection until start of recording.
WAIT	(RGB)	Provide ringback for call queuing for RAN trunk (default). MUS = Provide music for calls queuing for RAN trunk.
- MRT	0-511 0-127	Music route for RAN queuing. For Option 11C. MRT is only prompted for RAN route with WAIT = MUS.
BDCT	YES	Allow broadcast capability for this route. (NO) = Deny broadcast capability for this route (default).
- TITH	(0)-300	Waiting Threshold in seconds. Default value of zero means no threshold applies.
- NCTH	(0)-100	Number of Calls Waiting Threshold. Default value of zero means no threshold applies.
ASUP	(NO)	Do not return answer supervision (default). YES = Return answer supervision. CO = Return answer supervision only if originator is a Central Office trunk.
ACOD	x...x	Access Code for the trunk route. The Access Code must not conflict with the numbering plan. ACOD can be four digits, or seven digits with DNXP package 150 equipped.

LD 14 – Define new RAN Trunk.

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RAN	Recorded Announcement trunk data block.
TN	l s c u c u	Terminal Number For Option 11C.
...		
XTRK	a...a	Extended Trunk. To specify hardware, according to the RAN mode defined in LD 16, refer to Table 135.
RTMB	0-511 1-254 0-127 1-254	Route number and Member number. For Option 11C.
- CONN	(4)-48	Define the maximum number of broadcast connections allowed for this trunk. ***Note:*** *CONN is only prompted for associated RAN route with broadcasting allowed (BDCT=YES in LD 16).*

Note: The following feature implementation is applicable to customers using the Meridian Integrated Recorded Announcement (MIRAN) card.

LD 16 – Define Continuous RAN route with Meridian Integrated RAN (MIRAN).

Prompt	Response	Description
REQ	NEW, CHG	New, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-511 0-127	Route Number. For Option 11C.
TKTP	RAN	Recorded Announcement trunk type.
RTYP	MCON	Continuous Multi-channel.

- LGTH	4-(60)-7200	Maximum message length in seconds. This is only prompted for the continuous mode multichannel, the level start/stop multichannel and the pulse start/stop multichannel.
- GRD	(IDLE)	Ground signal from RAN indicates that machine is idle (default). PLAY = Ground signal from RAN indicates that machine is playing.
REP	1-15	Repetitions of recorded announcements.
POST	aaa	RAN Post announcement treatment where: ATT = Route to attendant after maximum repetitions DIS = Disconnect after maximum repetitions.
STRT	aaa	Start arrangement where: IMM = Immediately connect call to recording. DDL = Delay call connection until start of recording.
WAIT	(RGB)	Provide ringback for call queuing for RAN trunk (default). MUS = Provide music for calls queuing for RAN trunk.
- MRT	0-511 0-127	Music route for RAN queuing. For Option 11C. MRT is only prompted for RAN route with WAIT = MUS.
BDCT	YES	Allow broadcast capability for this route. (NO) = Deny broadcast capability for this route (default).
ASUP	(NO)	Do not return answer supervision (default). YES = Return answer supervision. CO = Return answer supervision only if originator is a Central Office trunk.
ACOD	x...x	Access Code for the trunk route. The Access Code must not conflict with the numbering plan. ACOD can be four digits, or seven digits with DNXP package 150 equipped.

LD 16 – Define Immediate Start/Stop RAN route with Meridian Integrated RAN (MIRAN).

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-511 0-127	Route Number. For Option 11C.
TKTP	RAN	Recorded Announcement trunk type.
RTYP	MLVL	Level start/stop multichannel recording devices for RAN trunks.
- LGTH	4-(60)-7200	Maximum message length in seconds. This is only prompted for the continuous mode multichannel, the level start/stop multichannel and the pulse start/stop multichannel.
- GRD	(IDLE)	Ground signal from RAN indicates that machine is idle (default). PLAY = Ground signal from RAN indicates that machine is playing.
REP	1-15	Repetitions of recorded announcements.
POST	aaa	Post RAN treatment where: ATT = Route to attendant after maximum repetitions DIS = Disconnect after maximum repetitions.
STRT	IMM	Immediately connect call to recording.
WAIT	(RGB)	Provide ringback for call queuing for RAN trunk (default). MUS = Provide music for calls queuing for RAN trunk.
- MRT	0-511 0 -127	Music route for RAN queuing. For Option 11C. MRT is only prompted for RAN route with WAIT = MUS.
BDCT	YES	Allow broadcast capability for this route. (NO) = Deny broadcast capability for this route (default).

ASUP	(NO)	Do not return answer supervision (default). YES = Return answer supervision. CO = Return answer supervision only if originator is a Central Office trunk.
ACOD	x...x	Access Code for the trunk route. The Access Code must not conflict with the numbering plan. ACOD can be four digits, or seven digits with DNXP package 150 equipped.

LD 16 – Define Delay Dial Start/Stop RAN route with Meridian Integrated RAN (MIRAN).

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number.
ROUT	0-511 0-127	Route Number. For Option 11C.
TKTP	RAN	Recorded Announcement trunk type.
RTYP	MLVL	Level start/stop multichannel recording devices for RAN trunks.
- LGTH	4-(60)-7200	Maximum message length in seconds. This is only prompted for the continuous mode multichannel, the level start/stop multichannel and the pulse start/stop multichannel.
- GRD	(IDLE)	Ground signal from RAN indicates that machine is idle (default). PLAY = Ground signal from RAN indicates that machine is playing.
REP	1-15	Repetitions of recorded announcements.
POST	aaa	RAN Post announcement treatment where: ATT = Route to attendant after maximum repetitions DIS = Disconnect after maximum repetitions.
STRT	DDL	Delay call connection until start of recording.

WAIT	(RGB)	Provide ringback for call queuing for RAN trunk (default). MUS = Provide music for calls queuing for RAN trunk.
- MRT	0-511 0-127	Music route for RAN queuing. For Option 11C. MRT is only prompted for RAN route with WAIT = MUS.
BDCT	YES	Allow broadcast capability for this route. (NO) = Deny broadcast capability for this route (default).
- TITH	(0)-300	Waiting Time Threshold in seconds. The default value of (0) means no threshold applies. TITH is only prompted when BDCT = YES and STRT = DDL.
- NCTH	(0)-100	Number of Calls Waiting Threshold. Default value of zero means no threshold applies. NCTH is only prompted when BDCT = YES and STRT = DDL.
ASUP	(NO)	Do not return answer supervision (default). YES = Return answer supervision. CO = Return answer supervision only if originator is a Central Office trunk.
ACOD	x...x	Access Code for the trunk route. The Access Code must not conflict with the numbering plan. ACOD can be four digits, or seven digits with DNXP package 150 equipped.

LD 14 – Define a RAN trunk.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RAN	Recorded Announcement trunk data block.
TN	l s c u c u	Terminal Number For Option 11C.
XTRK	EXUT	Enhanced Extended Universal Trunk card. To use the new MIRAN card, the XTRK prompt must be set to EXUT.
...		
RTMB	0-511 1-510 0-127 1-510	Route number and Member number. For Option 11C.
- CONN	(4)-48	Maximum number of broadcast connections allowed for this trunk. ***Note:*** *CONN is only prompted for associated RAN route with broadcasting allowed (BDCT=YES in LD 16).* ***Note:*** *The CONN prompt defines the maximum number of broadcast connections allowed for a RAN trunk at any given time. As an example, if sixteen is configured, then the physical broadcasting trunk may broadcast up to sixteen callers at one time.*

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	20.1x
Networking:	Yes

Recorded Announcement for Calls Diverted to External Trunks

Recorded Announcement for Calls Diverted to External Trunks (RANX) provides an optional recorded announcement when the call is being forwarded to external Public Exchange/Central Office (CO) (DTI, DTI2, PRI2, PRI, analog, and BRI connected to AXE-10 or EuroISDN) routes. The announcement notifies the calling party that call forwarding is taking place and the call may take longer than usual to set up. The delay depends on the required signaling to reach the destination party.

The message is given if the outgoing route is supported by the RANX feature. The calling party receives Recorded Announcement (RAN) treatment until either the message is finished or until an answer is received from the external CO trunk.

This feature operates on a route basis and is controlled by the RANX prompt in LD 16 for calls diverted to external trunks.

Operating parameters

The RANX feature is supported only for external CO routes. The corresponding RANX prompt in LD 16 appears when the trunk type is:

- A Central Office Trunk (COT) and is not configured as a Radio Paging (RPA) trunk. The trunk has to be configured as an outgoing or outgoing/incoming trunk. The feature is not applicable for trunks configured solely for data traffic.
- A Direct Inward Dialing (DID) trunk configured as an outgoing or outgoing/incoming trunk. The feature is not applicable for trunks configured solely for data traffic.

The RANX feature requires a Recorded Announcement (RAN) machine in the same node as the outgoing route.

If the RAN trunk is configured with supervision, the calling party who gets connected to the RAN trunk will be charged from the time when the answer signal is sent.

This feature is supported network wide in a Meridian Customer Defined Network (MCDN) environment; no other private network protocols are supported.

Since Digital Private Network Signaling System (DPNSS1) does not support any information concerning redirection, this feature is not supported in a DPNSS1 network.

If a telephone is forwarded to a trunk configured with the RANX feature and the party that calls the extension is supposed to dial additional digits this is not always possible. If the calling party dials digits when being provided with RAN, those digits will be lost since they are not buffered.

If the outgoing trunk, or the set from which the call originated, times out during the RAN it is not possible to dial additional digits after the RAN is terminated.

If a Direct Inward System Access (DISA) call is forwarded to a route with the RANX feature configured, and the RAN message that is provided to the calling party is longer than the duration of the EOD timer, the RAN will be interrupted when the End of Dial (EOD) timer expires. When the EOD timer expires the call is considered as established.

It is recommended not to use this feature if the outgoing external CO route is not configured with answer supervision.

Feature interactions

Call Forward All Calls
Call Forward Busy
Call Forward No Answer
Hunt

RANX is activated if the call is forwarded to an outgoing external CO trunk with the RANX feature active.

Call Forward, Internal Calls

The RANX feature supports call forward to an outgoing external Central Office (CO) trunk if the trunk has the RANX flag set and is located in a node with a RAN trunk.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

The Recorded Announcement for Calls Diverted to External Trunks feature is not supported in a DPNSS1 UDP network.

Expensive Route Warning Tone

If the calling party is being forwarded to a route with the RANX feature and the Expensive Route Warning Tone feature configured, the Expensive Route Warning Tone will be heard prior to the recorded announcement.

Malicious Call Trace Idle

DID calls to a busy Recorded Announcement (RAN) trunk group are queued and receive ring-back tone. A Multifrequency Compelled IDLE signal is returned.

Network Call Forward

The RANX feature supports call forward to an outgoing external CO route if the route has RANX configured and is located in a node with a RAN trunk. The originating party and the forwarded set may be in different nodes in the MCDN network.

Phantom Terminal Numbers (TNs)

If a Phantom TN is forwarded to an external outgoing CO route and the RANX feature is configured for this route, the calling party that is forwarded due to the Phantom TN feature will be provided with a recorded announcement.

Feature packaging

There is no new software package required; however, the existing Recorded Announcement (RAN) package 7; and the Intercept Treatment (INTR) package 11 must be provisioned to activate this feature.

Feature implementation

LD 16 – Two new prompts have been introduced to this overlay by this feature: RANX and RANR.

Prompt	**Response**	**Description**
REQ	NEW CHG	New, or change.
TYPE	RDB	Route data block.
...		
FORM	aaa	Signaling format.
ICOG	IAO OGT	Incoming and/or outgoing trunk.
RANX		RAN for calls diverted to external trunks. This prompt appears if TKTP = COT or DID, RPA = NO, DSEL = (VCE or VOD) and ICOG = (IAO or OGT).
	(NO)	RANX is set to no; RAN is not requested when a call is forwarded to this route.
	YES	RANX is set to yes; RAN is requested when a call is forwarded to this route.
	<CR>	The RANX value is not changed.
RANR	0-511	Appears only if the response to RANX is YES. The route number for the desired RAN is selected.
	0-127	Route number for the Option 11.

Feature operation

No specific operating procedures are required to use this feature.

Recorded Overflow Announcement

Recorded Overflow Announcement (ROA) allows delayed calls to the attendant to be connected to a recorded announcement notifying the calling party of the delay. A second recorded message can also be provided to the calling party repeatedly until an attendant answers the call.

A call that is waiting in the queue receives the first recorded message after the expiration of a timer (T1). After the message is given, the call returns to the attendant queue. While the call is in the waiting state, it can be connected either to Music (MUS), Ringback tone (RGB), or Silence (SIL).

If a second recorded announcement is specified, the call receives the message upon expiration of a second timer (T2). After the second message is given, the call is placed in the attendant queue again. There is no limit to the number of times a call can be given the second recorded message.

Operating parameters

Recorded Overflow Announcement (ROA) treatment is provided to call types assigned to Incoming Call Indicator (ICI) keys on the Attendant Console.

A maximum of 20 ICI keys can be assigned to receive Recorded Overflow Announcement (ROA) treatment.

The delay time thresholds for the first and second recorded announcements (T1 and T2) are assigned in LD 15. The thresholds shown in Table 139 can be defined for these timers.

Table 139
Delay time thresholds

	Thresholds		
	Minimum	**Default**	**Maximum**
T1	0 seconds	20 seconds	2,044 seconds
T2	0 seconds	40 seconds	2,044 seconds

Loop start trunks do not provide disconnect supervision and are not recommended for use with the ROA feature. A call on a loop start trunk that is abandoned after the recorded message is given must be manually cleared by the attendant.

ROA is not provided on release link trunks from Centralized Attendant Service (CAS) remote locations.

When the CAS feature is activated at a remote PBX, the ROA feature is inactive at the remote site.

If music is required, the Music (MUS) package 44 must be equipped. Music can be provided after the first and second Recorded Announcement (RAN). A customer provided music source is required, connected through a Music trunk. Music is provided to delayed calls through a conference circuit pack in a listen-only mode. The music source provided by the customer must be compatible with the RAN trunk card.

Prior to X11 Release 15, Music (MUS) and Recorded Announcement (RAN) could not share the same trunk card.

Private Lines are not eligible for ROA.

ROA is only provided for call types assigned to Incoming Call Indicator (ICI) keys. The following call types are eligible, if related ICI keys are assigned:

— Trunk routes

- LDN 0 through LDN 3
- Dial 0
- Dial 0 Fully Restricted
- Intercept Treatment (INTR)
- Call Forward Busy
- Call Forward No Answer
- Message Waiting (MW)
- Lockout, and
- Station Category Indication (SCI).

Feature interactions

Automatic Call Distribution (ACD)

The RAN route used for ROA can be the same route that is used for ACD and Intercept Treatment.

Night Service

The ROA feature is inactive when the system is in Night Service.

Feature packaging

Recorded Overflow Announcement (ROA) package 36 requires Recorded Announcement (RAN) package 7.

Feature implementation

LD 16 – Add or change Recorded Announcement (RAN) trunk route.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number.
ROUT	0-511	Route number.
TKTP	RAN	RAN trunks.

RTYP	CAP	Code-a-Phone recording device. Software allows announcements of up to 320 seconds in length in X11 Release 14, or 608 seconds in X11 Release 15.
	AUD	Audichron recording device (required when connecting to a Universal Trunk Card). Software allows announcements of up to 64 seconds.
	CK2	Cook Electric recording device. Software allows announcements of up to 64 seconds.
	DGT	Digital Recorders 213300 & 213400. Software allows announcements of up to 256 seconds on X11 Release 15 and later.
	CON	NT7M series digital recorders. Software allows announcements of up to 608 seconds on X11 Release 15 and later.
REP	1-15	Number of times the announcement repeats during each connection.
POST	ATT	Call is routed to attendant after specified number of repetitions (applies to Direct Inward Dial [DID] calls on Intercept).
	DIS	RAN is removed after a specified number of repetitions (call is kept in Automatic Call Distribution queue).
STRT	IMM	Call connects immediately to announcement.
	DDL	Call connects to announcement at the start of announcement.
ASUP	YES	Return Answer Supervision by RAN to originator. ASUP=NO (Default) ***Note:*** ASUP must be set to YES to allow the following options in LD 15 (at the WAIT prompt): Caller hears Ringback (RGB), Music (MUS), or Silence (SIL) while waiting.
ACOD	xxx...x	Trunk route access code.

Note: All RAN route members must be removed before the route can be removed.

LD 14 – Add or change Recorded Announcement (RAN) trunk.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	RAN	RAN trunk data block.
TN	l s c u	Terminal Number.
CUST	0-99	Customer Number (prompted if REQ = NEW).
RTMB	xxx yyy	Route and member number, where: xxx = 0-511, and yyy = 1-254.

LD 15 – Enable a Recorded Announcement (RAN) route for the customer.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB ROA	Customer Data Block. Release 21 gate opener.
CUST	0-99	Customer number.
OPT	(ROX), ROI	Recorded Overflow (excluded) included.
- FRRT	xxx	Route number for the first recorded announcement.
- FRT	0-(20)-2044	Time in seconds before the first announcement plays.
- SRRT	xxx	Route number for the second recorded announcement.
- SRT	0-(40)-2044	Time in seconds before second announcement plays.
- WAIT	RGB, MUS, SIL	Caller hears Ringback (RGB), Music (MUS), or Silence (SIL) while waiting.
- - MURT	xxx	Route Number for Music route if WAIT = MUS.
- RICI	xx . .xx . .xx	Incoming Call Indicator (ICI) key numbers eligible for ROA.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	All
Networking:	Yes

Recorded Telephone Dictation

This feature provides dial access to customer-supplied dictation equipment. Operation of the equipment can be either voice or dial controlled. The actual controls vary with the type of dictation equipment used.

To access the dictation equipment, the user dials the access code assigned to the dictation route. Access to the route is controlled by Trunk Group Access Restrictions (TGARs).

Operating parameters

Each recorded dictation unit requires a separate trunk route.

Feature interactions

Multi-Party Operations

Users of analog (500/2500 type) telephones cannot make a consultation call while connected to a dictation trunk.

Conference

Recorded Telephone Dictation trunks cannot be used in a conference call.

Feature packaging

Recorded Telephone Dictation is included in base X11 system software.

Feature implementation

LD 16 – Add or change a trunk route for the Recorded Telephone Dictation feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	0-99 0-31	Customer number. For Option 11C.
ROUT	0-511 0-127	Route number. For Option 11C.
TKTP	DIC	Recorded Telephone Dictation trunk route.
ICOG	OGT	Outgoing trunk route.
ACOD	xxx...x	Directory Number (DN) to dial to access the dictation device.

LD 14 – Add or change a trunk for the Recorded Telephone Dictation feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
TN	l s c u c u	Terminal Number. For Option 11C.
CUST	0-99 0-31	Customer number. For Option 11C.
RTMB	rrr mm	Route and member number.
SIGL	aaa	Trunk signaling.
STRO	aaa	Outgoing start arrangement.
SUPN	(NO) YES	Answer and disconnect supervision (not) required.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	10.10C
Networking:	No

Recovery on Misoperation of Attendant Console

The Recovery of Misoperation on the Attendant Console feature provides a safeguard in the Meridian 1 software that prevents calls from being inadvertently disconnected.

Operating parameters

For Centralized Attendant Service, misoperation of the Attendant Console at the main node cannot be prevented.

Feature interactions

Call Forward All Calls
Call Forward Busy
Call Forward by Call Type
Call Forward External Deny
Call Forward, Internal Calls
Call Forward No Answer
Call Forward No Answer, Second Level
Hunting

These features take precedence over the Recovery of Misoperation feature.

Electronic Switched Network

If the attendant dials an incomplete Electronic Switched Network (ESN) number as a destination, pressing the Release key or another loop key is ignored. The attendant can dial more digits as long as the interdigit timer has not timed out. To dial to another number, the attendant must first press the Release Destination key to release the destination.

Music

Music on Hold, if allowed, is applied to calls put on hold due to the Autohold on the loop key option.

Recorded Announcement

If a recorded announcement is given to the destination side that has been intercepted, the connection to the destination side is considered as invalid. Therefore, if the attendant tries to extend the source to the destination using the Release key or another loop key, the operation is ignored. The attendant must first press the Release Destination key to release the destination, and then extend the call to the source. If the Hold key is pressed, the source party is put on hold and the recorded announcement is disconnected on the destination side.

Through Dialing

If an attendant dials a trunk access code and then presses the Release key or another loop key, the station on the source side and the trunk on the destination side are connected and released from the console. The source can then dial the remaining digits to access an outside destination. The Hold key is ignored.

Feature packaging

Recovery on Misoperation of Attendant Console is included in base X11 system software.

Feature implementation

LD 15 – Activate Recovery on Misoperation of Attendant Console.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
CUST	0-99 0-31	Customer number. For Option 11C.
OPT	(AHD) AHA	Autohold on loop Key (denied) allowed.
	(REA) RED	Release on Exclusion (allowed) denied.

Feature operation

This section describes how the feature works in each of the following cases:

— Misoperation of Release key and loop keys

— Misoperation of Autohold on loop key

— Misoperation of Release Source key and Release Destination key.

Misoperation of Release key and loop keys

In the following cases, pressing the Release key or the loop key is ignored:

— Extending a call to a vacant number

— Extending a call to restricted station or trunk

— Extending to a station restricted by Trunk Barring

Note: Intercept treatment is returned for the above conditions.

— Extending to a partially-dialed number

— Extending a network-blocked call

— Extending a station in the Do Not Disturb mode

— Extending to a station in the Make Set Busy mode

— Extending to a station in the Maintenance-busy state

— Extending to a station in the Line Lockout state

— Extending to a busy extension without Camp-on or Call Waiting

— Extending to a station restricted by Trunk-to-Trunk Connection Restriction

— Releasing from a conference connection – The attendant is prevented from releasing a conference connection, established on the source side, by pressing the Release key or a loop key in the following cases:

- if there is no destination. Pressing either the Release key or a loop key places the active loop on hold rather than releasing it. The conference can be released by pressing the Release Source key.
- if the attempt to extend the call to the destination was not successful. The conference can be released by pressing the Release Destination key.
- if there is another party already connected as a destination. Pressing the Hold key, Release key or another loop key puts the active loop on hold, rather than releasing it. The destination side can be released by pressing the Release Destination key. The source side can be released by pressing the Release Source key. If an established conference connection cannot be released due to Trunk-to-Trunk Connection Restriction, pressing the Release Source key causes the conference to be released from the console and the trunks disconnected.

Note: Busy tone or overflow tone is returned for the above conditions.

Misoperation of Autohold on the loop key

On a console that is equipped with the Autohold on loop key option, if the attendant is on a call that has terminated properly and presses the loop key while switching to another call, the active loop is placed on hold rather than being released. Besides preventing the inadvertent release of the caller, this option allows the attendant to toggle between any number of held calls by having to press only one key. If the attendant is on a call that cannot be terminated properly, pressing another loop key releases the destination side and puts the source side on hold.

In the following cases, pressing the Release key or the loop key places the call on hold rather than releasing it.

— Extending to a busy extension without Camp-on or Call Waiting, or

— Extending to a station restricted by Trunk-to-Trunk Connection Restriction.

Misoperation of the Release Source/Release Destination key

This option allows the Meridian 1 system to ignore the pressing of the Release Source or Release Destination key, preventing the release of either the excluded source or destination party, or a conference call connection. The source or destination party involved in a talking connection with the attendant may still be released by pressing the Release Source or Release Destination key, as appropriate. In a lockout situation, where both source and destination parties are excluded, the attendant may use either the Release Source or Release Destination key to disconnect both parties, since the attendant is not able to re-enter the connection.

Reference Clock Switching

This product improvement allows a Clock Controller reference to automatically switch to another tracking reference if the reference goes into a non-acceptable state (the Clock Controller can track on its primary reference, secondary reference, or be in free run). A non-acceptable state is considered as one of the following:

— The reference loop is disabled.

— For 2 Mbps Primary Rate Interface (PRI2), one of the following group 2 errors is detected on the reference loop:

- The far end is in Out-of-service state
- The far end has lost Multiframe Alignment Signal
- Alarm Indication Signal is sent
- Loss of Frame Alignment, and
- Loss of Multiframe Alignment.

— For DTI2, if the reference loop is in Out-of-service (OOS) grade of service, or if the reference loop is in No New Call state, if the OOS is inhibited.

Clock references are supplied to the Clock Controller by the DTI2/PRI2 pack during tracking mode. As mentioned, the Clock Controller can track on its primary reference, secondary reference, or be in free run. If tracking on primary reference and a non-acceptable state is reached, the Clock Controller switches off primary reference and tracks on secondary reference, if it is in an acceptable state, or goes into free run. While tracking in secondary reference, the Clock Controller makes regular periodic checks, at the Clock Controller Audit Rate (CCAR), to determine whether tracking can resume on the primary reference. When the primary reference returns into acceptable state, tracking on primary reference resumes during the next Clock Controller audit.

The same processing occurs if the Clock Controller is tracking on secondary reference, and it goes into a non-acceptable state. It goes into primary reference, if in acceptable state, or free run.

When tracking in free run and an non-acceptable state is encountered, the Clock Controller will first attempt to track on primary state, if in an acceptable state, and then on secondary state. The free run tracking is controlled by a free run guard timer, which is started as soon as tracking begins in free run. As soon as this timer runs out, tracking is attempted on the primary reference and then on the secondary reference. If both are still in a non-acceptable state, tracking continues in free run and the free run guard timer is restarted. If the free run guard timer is not configured, the attempt to switch over to primary or secondary reference is made only as part of the Clock Controller check for an acceptable state on the primary and secondary references.

When the Clock Controller switches from one reference to another, a small delay occurs due to the loop status update and the switching process. During this delay, the reference is given by the Clock Controller to itself in hardware free run state.

Operating parameters

Clock Controller card QPC775, and circuit packs QPC915 and QPC536 (2 Mbps Digital Trunk Interface), and/or NT8D72AA (PRI2).

Feature interactions

There are no interactions with other features.

Feature packaging

Reference Clock Switching requires the following packages:

— International Supplementary Features (SUPP) package 131

— 1.5 Mbps Digital Trunk Interface (PBXI) package 75

— one or both of 2 Mbps Digital Trunk Interface (DTI2) package 129 and 2 Mbps Primary Rate Interface (PRI2) package 154

Feature implementation

LD 60 – Enable automatic switch over of system clock sources on the Clock Controller.

Command	Description
...	
EREF	Enable automatic switch over of system clocks. Enable automatic switch over of primary and secondary reference clocks. Also enables recovery of primary or secondary clocks when loops associated with these clocks are automatically enabled.

LD 73 – Enable fast clock switching.

Prompt	Response	Description
...		
CCGD	0-(15)-1440	Clock Controller free run Guard time (in minutes).
CCAR	0-(15)	Clock Controller Audit Rate. The time, in minutes, between normal CC audits. Only programmable on units equipped with 2.0 Mbps DTI/PRI.
EFCS	(NO) YES	Enable Fast Clock Switching.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	15
Networking:	No

Remote Call Forward

Remote Call Forward (RCFW) allows a telephone user to program Call Forward from a remote telephone. With Remote Call Forward (RCFW) enabled, forwarding DNs can be defined and Call Forward All Calls can be activated from within the Meridian 1 or outside the local switch. The Remote Call Forward (RCFW) feature is password protected.

The Station Control Password (SCPW) is required to program Remote Call Forward. Entering a password length of 0 disables the password control for both Electronic Lock and RCFW.

Operating parameters

RCFW requires the following:

— set the password length in LD 15, at the SCPL prompt

— add passwords in LD 10 and LD 11, at the SCPW prompt

— allow Call Forward All Calls in LD 10 and LD 11, and

— define Remote Call Forward Activate (RCFA), Deactivate (RCFD), and Verify (RCFV) Flexible Feature Codes (FFC) in LD 57.

To activate RCFW from outside of the local switch, you must use the Direct Inward System Access (DISA) DN. The telephone's Prime DN is associated with the RCFW password for added security. Also, RCFW can activate or deactivate Call Forward on a telephone, and verify the same feature on a telephone.

Changes to the Station Control Password length do not take effect until after a data dump and SYSLOAD.

If there are two telephones with the same Prime DN, it is recommended that only one of them have a Station Control Password. With RCFW, it is possible that two telephones could have the same password assigned. With the same password, they could control each other's security. For the same reason, the Secondary DN for an Automatic Call Distribution (ACD) telephone should not appear as a Prime DN on another telephone.

A unique number code must be programmed for each of the FFC functions relating to RCFW: Remote Call Forward Activate (RCFA), Remote Call Forward Deactivate (RCFD), and Remote Call Forward Verify (RCFV). You can change the RCFW Directory Number (DN) from your own telephone or from a telephone remote from the switch.

RCFW is not supported for ACD telephones.

Feature interactions

Attendant Administration

Attendant Administration does not support the telephone programming associated with Remote Call Forward.

Call Forward, Internal Calls

Remote CFW Activate (RCFA), Remote CFW Deactivate (RCFD), and Remote CFW Verify (RCFV) FFCs can be used only to access CFW All Calls; they cannot be used to access Internal Call Forward.

China – Flexible Feature Codes - Outgoing Call Barring

Activation of CFW to a barred DN by Remote Call Forward will be permitted, since the user has had to dial the Station Control Password, which could also have been used to deactivate OCB.

Multiple Appearance Directory Number

With a Multiple Appearance Directory Number (DN) and both sets having a Station Control Password (SCPW), Remote Call Forward may not operate as intended (that is, if Call Forward has been activated using the Remote Call Forward feature, Call Forward remains activated when an attempt to deactivate it is made from the set on which it is active).

Phantom Terminal Numbers (TNs)

If Remote Call Forward is to be used in conjunction with a Phantom TN, the Phantom TNs must be configured with the Call Forward All Calls (CFW) feature.

Preventing Reciprocal Call Forward

This modification applies to Remote Call Forward.

Set-Based Administration Enhancements

A set may be remote call forwarded while someone is actively logged into it with Set-Based Administration login.

2500 Telephone Features

When Flexible Feature Codes (FFC) package 139 is defined and active on your system, a telephone provisioned for Call Forward in LD 10 can also Call Forward All Calls from a remote internal DN.

Feature packaging

The following software packages are required to implement Remote Call Forward:

— Optional Features (OPTF) package 1

— Flexible Feature Codes (FFC) package 139, and

— Controlled Class of Service (CCOS) package 81.

In addition, the following software packages are required to implement RCFW on analog (500/2500 type) telephones:

— Special Service for 2500 (SS25) package 18, and

— 500 Set Dial Access to Features (SS5) package 73.

Feature implementation

LD 15 – Set the Station Control Password length.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB FFC	Customer Data Block. Release 21 gate opener.
CUST	0-99 0-31	Customer number. For Option 11C.
...		
- SCPL	0-8	Station control password length (0-8). Entering 0 disables the Remote Call Forward and the Electronic Lock features. ***Note:*** A data dump and SYSLOAD are required to implement a change in password length. Shorter passwords are filled with leading zeros. Passwords that are too long have the leading digits truncated.
- FFCS	YES	Change end of dialing digits in FFC.
- - STRL	1-3	Number of digits to indicate FFC end of a feature activation.
- - STRG	(#), xxx	1 to 3 digits to indicate FFC end of a feature entry.

LD 57 – Define Remote Call Forward FFCs.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FFC	Flexible Feature Codes.
FFCT	(NO) YES	FFC Confirmation Tone (optional).
CODE	RCFA	Remote Call Forward Activate.
RCFA	xx	RCFA code
CODE	RCFD	Remote Call Forward Deactivate.

RCFD	xx	RCFD code.
CODE	RCFV	Remote Call Forward Verify.
RCFV	xx	RCFV code.

LD 10 – Set the Station Control Password for analog (500/2500 type) telephones and allow Call Forward.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
SCPW	xxx...x	Station control password (0-8 digits as defined by prompt SCPL in LD 15).
	X	Entering X deletes the password.
FTR	CFW 4-(16)-23	Allow Call Forward and set forwarding DN length.

LD 11 – Set the Station Control Password for Meridian 1 proprietary telephones and allow Call Forward.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
SCPW	xxx...x	Station control password (0-8 digits as defined by prompt SCPL in LD 15).
	X	Entering X deletes the password.
KEY	xx CFW 4-(16)-23	Assign Call Forward key (xx) and set forwarding DN length.

Feature operation

From any telephone within the system, simply lift the handset and use the following procedures. From any telephone outside the system, first dial the Direct Inward System Access (DISA) number for your system, wait for dial tone, and dial any required passwords and Authorization Codes.

1 Dial the Remote Call Forward Activate FFC.

2 Dial the Station Control Password for the telephone to be forwarded.

3 Dial the Prime DN of the telephone to be forwarded.

4 Dial the number to which calls will be forwarded.

5 Dial the end-of-entry digit(s) (defined in LD 15), if these digits plus the number of digits in the forwarding DN are less than 24 digits. (If you do not dial the end-of-entry digits, the forwarding DN is saved but cannot be verified remotely.)

You will hear a confirmation tone after entering the main extension number, telling you that the password and extension match. You will hear a second special tone after dialing the end-of-entry digits, telling you that the procedure was successful. If you hear a fast busy signal, hang up and try again.

When entering the forwarding DN, you cannot enter more than 23 digits, including the end-of-entry digits. If you attempt to enter a 24th digit, you will hear an overflow tone.

If the forwarding DN plus the end-of-entry digits are not less than 24 digits, hang up after dialing the forwarding DN. The DN is saved but cannot be verified remotely.

To cancel Remote Call Forward:

1 Dial the Remote Call Forward Deactivate FFC.

2 Dial the Station Control Password for the telephone.

3 Dial the Prime DN of the telephone.

To verify Remote Call Forward:

1 Dial the Remote Call Forward Verify FFC.

2 Dial the Station Control Password for the telephone.

3. Dial the Prime DN of the telephone.
4. Dial the number to which calls should be forwarded.
5. Dial the end-of-entry digit(s).

If the number to which the telephone is forwarding calls does not match your entry in step 4, you will hear a fast busy signal. If the numbers do match, you will hear a confirmation tone after entering the forwarding number, provided the confirmation tone is enabled in LD 57.

When entering the forwarding DN, you cannot enter more than 23 digits, including the end-of-entry digits. If you attempt to enter a 24th digit, you will hear an overflow tone. You cannot use Remote Call Forward Verify for a forwarding DN that was entered without the end-of-entry digits because of too many digits.

Introduced in X11 Release:	All
Networking:	No

Remote Peripheral Equipment

The Remote Peripheral Equipment (RPE) feature allows the range of the multiplexed loop between common and peripheral equipment to be extended beyond the normal 14 m (50 ft.), to about 100 km (70 miles) using T1 carrier facilities. This carrier system must conform to the North American T1 specification to link the local and remote locations, and can consist of the following:

— 24-gauge wire pairs for applications in which the remote end is less than 2500 feet from the Meridian 1 common equipment

— a Digital carrier link (such as Northern Telecom LD-1), or

— a microwave radio link.

The Remote Peripheral Equipment (RPE) feature allows the peripheral equipment to be placed closer to the stations it serves, and increases the serving area of a single system.

Among the benefits are the following:

— normal attendant operation covering all locations

— elimination of TIE lines between locations

— uniform system features, and

— a fully integrated numbering plan.

For details regarding RPE, refer to Northern Telecom Publication *Remote Peripheral Equipment description, installation, and testing* (553-2601-200).

Operating parameters

Refer to *Remote Peripheral Equipment description, installation, and testing* (553-2601-200).

Feature interactions

Refer to *Remote Peripheral Equipment description, installation, and testing* (553-2601-200).

Feature packaging

Remote Peripheral Equipment (RPE) package 15 has no feature package dependencies.

Feature implementation

If an even-numbered Tone and Digit Switch (TDS), CONF, or MFSD loop (0, 48, 72, 150) is equipped, the succeeding odd-numbered loop (1, 49, 73, 151) cannot be assigned as a voice loop.

The Peripheral Buffer card switch must be set for quad density. After changes are made, the system must be initialized to activate the changes to the network loop in the database.

LD 17 – Add or change a voice/RPE loop(s).

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN CEQU	Configuration Record. Release 19 gate opener.
CEQU	(NO) YES	Allow changes to common equipment parameters.
- MPED	SD DD 4D	Maximum peripheral equipment density.
- TERM	xxx yyy	Single density local terminal loops. For nonenhanced networks: xxx = 0-79 yyy = 0-79 For enhanced networks: xxx = 0-159 yyy = 0-159

- REMO	xxx yyy	Single density remote terminal loops. For nonenhanced networks: xxx = 0-79 yyy = 0-79 For enhanced networks: xxx = 0-159 yyy = 0-159
- TERD	xxx yyy	Double density local terminal loops. For nonenhanced networks: xxx = 0-79 yyy = 0-79 For enhanced networks: xxx = 0-159 yyy = 0-159
- REMD	xxx yyy	Double density remote terminal loops. For nonenhanced networks: xxx = 0-79 yyy = 0-79 For enhanced networks: xxx = 0-159 yyy = 0-159
- TERQ	xxx yyy	Quad density local terminal loops. For nonenhanced networks: xxx = 0-79 yyy = 0-79 For enhanced networks: xxx = 0-159 yyy = 0-159
- REMQ	xxx yyy	Quad density remote terminal loops. For nonenhanced networks: xxx = 0-79 yyy = 0-79 For enhanced networks: xxx = 0-159 yyy = 0-159

Feature operation

No specific operating procedures are required to use this feature.

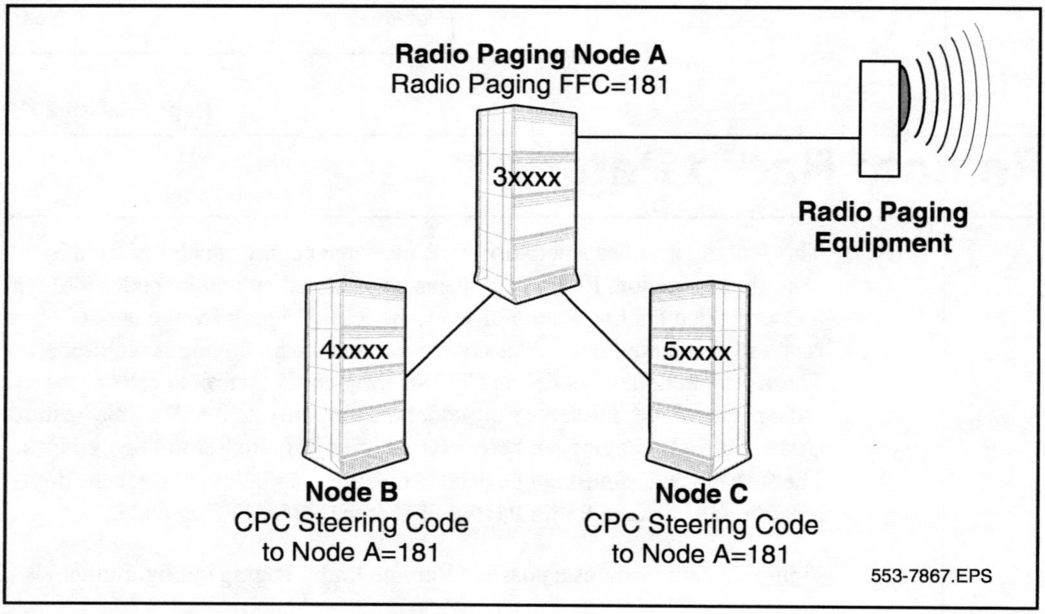
Introduced in X11 Release:	16.67G
Networking:	Yes

Remote Radio Paging

This feature provides a network-wide meet-me paging capability from a centralized location. Radio Paging can be accessed by remote nodes through a Coordinated Dialing Plan; however, the Radio Paging feature is not required at remote nodes unless post-selection Radio Paging is required. These remote nodes can define CDP steering codes that route calls to the radio paging node. These steering codes are the equivalent of Flexible Feature Codes for Radio Paging, and are referred to as Remote Radio Paging FFCs. The steering codes must not be deleted by digit manipulation, since the digits are interpreted as the Radio Paging FFC at the Radio Paging node.

Figure 77 demonstrates a possible Remote Radio Paging configuration.

Figure 77
A typical Remote Radio Paging configuration



Node A, which is equipped with the Remote Radio Paging feature, is referred to as the Radio Paging node. The Radio Paging FFC is defined as 181. At remote nodes B and C, steering codes of 181 have been defined to route calls to node A. To access Radio Paging from nodes B and C, a caller simply has to dial 181.

Post Selection Access to Remote Radio Paging

This feature allows the post selection operation of Radio Paging from all nodes in the network. For this functionality, all nodes must be equipped with the Remote Radio Paging feature. For post-selection access, Trunk Steering Codes (TSCs) and Distant Steering Codes (DSCs) are defined as Remote Radio Paging FFCs.

If a post-selection access is made to a set on the same node, the originally-called set must be either ringing or busy. If the originally-dialed set is on another node, it must be on an established call. In this latter case, the established call is disconnected before being routed to the radio paging node.

Post-selection access can be performed from 500/2500-type sets, SL-1 sets, Meridian 1000 series sets, Meridian digital sets, and Attendant Consoles.

Operating parameters

All DNs in the network must have the same fixed length.

The * and # symbols cannot be used as part of Radio Paging FFC.

Post Selection Access cannot be done using the single-digit access method.

Feature interactions

There are no interactions with other features.

Feature packaging

Controlled Class of Service (CCOS) package 81; Flexible Feature Codes (FFC) package 139; and Radio Paging (RPA) package 187.

Feature implementation

LD 87 – Create the Coordinated Dialing Plan TSCs and DSCs for remote nodes.

Prompt	Response	Description
...		
DSC	xxxx	Distant Steering Code. Respond with a four-digit value. The DSC must be identical to the Radio Paging FFC at the radio paging node.
- RRPA	(NO) YES	(Disable) enable Remote Radio Paging Access. Remote Radio Paging FFC is being used. Prompted if a CDP, TSC, or DSC is being changed.
TSC	xxxx	Trunk Steering code. Respond with a four-digit value. The TSC must be identical to the Radio Paging FFC at the radio paging node.

LD 11 – Assign the TSC or DSC steering code to the Radio Paging key on Meridian 1 proprietary telephones.

Prompt	Response	Description
...		
KEY	xx RPAG yyyy	Key number, Radio Paging, Route Access Code.

LD 12 – Assign the TSC or DSC steering code to the Radio Paging key on Attendant Consoles.

Prompt	Response	Description
...		
KEY	xx RPAG yyyy	Key number, Radio Paging, Route Access Code.

Feature operation

Feature operation is the same as for Radio Paging.

Restricted Call Transfer

This feature provides the Call Transfer Restricted (XFR) Class of Service for analog (500/2500 type) telephones. By assigning XFR Class of Service in LD 10, a Call Transfer attempt will not result in action. This is different from the Call Transfer Denied (XFD) Class of Service, which will route the call to the attendant when a transfer is attempted.

Operating parameters

The Three-party Service Allowed Class of Service, part of the Multiple-Party Operation feature, cannot be used together with the XFR Class of Service.

Feature interactions

There are no interactions with other features.

Feature packaging

Restricted Call Transfer is included in base X11 system software.

Feature implementation

LD 10 – Define two new options for an analog (500/2500 type) telephone.

Prompt	Response	Description
...		
CLS	XFR	Restrict call transfers and do not recall to attendant.

Feature operation

With XFR Class of Service assigned, a Call Transfer request will not result in action.

Restricted Direct Inward Dialing Class of Service

In order to restrict certain stations from receiving Direct Inward Dialing (DID) calls, the feature will either restrict DID (RDI) calls or unrestricted DID (UDI) calls. The RDI stations will fully restrict DID calls and whereas non-DID calls will be treated according to their normal Class of Service.

Operating parameters

The Central Office must be equipped to handle the special signaling requirements associated with the Restricted DID Class of Service feature described above.

The Restricted DID Class of Service feature is not available on 1.5 Mbps digital trunks or Japanese Digital Multiplex Interface (DMI) trunks.

Attendant Administration of the Restricted DID Class of Service is not available.

Feature interactions

Class of Service Restrictions

The Restricted DID Class of Service feature changes the access restrictions for telephone sets which have the feature enabled. These sets are treated as fully-restricted with respect to direct calls from DID trunks.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 10 – Define two new options for a analog (500/2500 type) telephones.

Prompt	Response	Description
...		
CLS	(UDI) RDI	This station (is not) is restricted from receiving direct DID calls.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	10.10B
Networking:	No

Reverse Dial on Routes and Telephones

This feature is used to allow a customer to define their dialpulse format as one of the following:

- regular dial format
- reverse dial format, or
- N+1 dial format.

The feature can be allowed or disallowed on either a route or on all telephones, on a customer basis, by associating a tone table (refer to 553-2201-310) with the route or customer, and setting the reverse dial format in the tone table as required.

Both the "*" and "#" are handled in the same manner as it exists in the regular format. Regular dial format is the default for the feature.

Operating parameters

The feature is supported for Central Office (CO), Foreign Exchange (FEX), Wide Area Telephone Service (WATS), TIE, and Direct Inward Dialing (DID) routes only. Internal Meridian 1 calls are unaffected, except when the feature applies to customers.

Feature interactions

There are no interactions with other features.

Feature packaging

Flexible Tones and Cadences (FTC) package 125.

Feature implementation

LD 56 – Modify or change customer's tone and ringing parameters:

Prompt	Response	Comment
...		
RDVL	(0)	No Reverse Dial format.
	1	Reverse Dial format 1 selected.
	2	Reverse Dial format 2 selected.

Feature operation

No specific operating procedures are required to use this feature.

Ring Again

Ring Again gives you the opportunity, after encountering a busy Directory Number (DN), to ring the DN again when it becomes free. If a dialed DN is busy, or if all the trunks are busy, pressing the Ring Again key asks the system to monitor the dialed DN or trunk. When it becomes available, the system notifies you. The call is automatically dialed again when you press the Ring Again key a second time.

When the system alerts you to Ring Again, you have a limited amount of time to respond. Analog (500/2500 type) telephones have six seconds, while Meridian 1 proprietary telephones have 30 seconds.

Operating parameters

A key/lamp pair must be assigned to Meridian 1 proprietary telephones for Ring Again. M3000 and M2317 telephones access Ring Again with a softkey.

Several people can activate Ring Again against the same DN while it is busy. When the DN becomes free, the system notifies the first person in line.

For analog (500/2500 type) telephones, a Special Prefix (SPRE) or Flexible Feature Code (FFC) may be used.

Feature interactions

Attendant Blocking of Directory Number

It is possible to activate Ring Again towards a DN that is blocked due to the Attendant Blocking of DN feature.

Attendant Overflow Position

If Ring Again is activated against the Attendant Overflow Position (AOP) DN, notification is given to the originator when the telephone becomes idle. An AOP call, however, takes precedence over Ring Again notification on the AOP DN when the AOP DN becomes free.

Automatic Set Relocation

If Ring Again is active when a telephone is relocated, the feature is deactivated.

Basic/Network Alternate Route Selection (BARS/NARS)

If the system is equipped with BARS or NARS, the Ring Again feature is used with the Call Back Queuing option to queue for outgoing trunks.

Call Forward/Hunt Override Via Flexible Feature Code

Using the Ring Again feature is possible after using the Call Forward/Hunt Override FFC and encountering a busy signal. Ring Again can be placed against the set for which the Call Forward/Hunt Override FFC was used (i.e., the set with CFW active should be rung by the Ring Again feature).

Call Waiting

The user is notified that a previously busy line is free only when both the original call and the waiting call have disconnected.

Charge Account and Calling Party Number

When Ring Again is activated, no charge record is generated, but the information is stored for future use. If Ring Again is canceled before a trunk is seized, the charge number is deleted and no record is produced. If a trunk is seized later by Ring Again, the charge record is generated in the usual manner. The use of Ring Again with Charge Account ties up system resources because an auxiliary call register must be maintained in the Ring Again queue.

China – Flexible Feature Codes - Outgoing Call Barring

Ring Again cannot be activated after a call is barred by Outgoing Call Barring. Sets with display will not offer Ring Again.

Conference

This feature cannot be activated during a conference call.

Dial Access to Group Calls
Group Call

Ring Again cannot be applied to Dial Access to Group Calls or Group Call.

Group Hunt

Ring Again will not be supported.

Idle Extension Notification

During the time that an extension is supervised or temporarily blocked from receiving calls due to the Idle Extension Notification feature, it is possible to activate Ring Again towards that extension. It is also possible to request for Idle Extension Notification on an extension that is supervised for Ring Again. When the extension becomes idle, the Idle Extension Notification will be served first.

ISDN QSIG/EuroISDN Call Completion

Analog (500/2500 type) sets can have only one Call Completion to Busy Subscriber request at a given time. Meridian 1 proprietary sets can make Ring Again requests based on the number of Ring Again keys programmed on a set.

Multi-Party Operations

When a TSA Class of Service analog (500/2500 type) telephone with a call on hold encounters Busy Tone, Ring Again is not possible.

Network Intercom

Hot Line calls terminating on a busy key become normal calls. Hence, they may use the Ring Again feature under normal circumstances.

On Hold on Loudspeaker

Ring Again can be applied to a busy loudspeaker DN.

Override
Override, Enhanced

Ring Again is the only other feature currently available once a busy telephone has been encountered. Ring Again is not allowed on an analog (500/2500 type) telephone making a Multi-Party Operations consultation call.

Feature packaging

Ring Again is included in Optional Features (OPTF) package 1 and has no feature package dependencies.

Feature implementation

LD 10 – Add or change Ring Again for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	(XRD) XRA	Ring Again is (denied) or allowed.

LD 11 – Add or change Ring Again for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx RGA	Ring Again key, where: xx = key number (must be key 27 for M2317 or M3000).

Feature operation

Ring Again is slightly different for each telephone type. Be sure to follow the correct operating instructions.

Meridian 1 proprietary telephones

To activate Ring Again after hearing a busy signal:

1. Press **Ring Again**.
2. Hang up, or press **Rls**.
3. When you hear the Ring Again tone, lift the handset or select a free **DN**.
4. Press **Ring Again**. The number is automatically dialed.

To cancel Ring Again:

— Press **Ring Again** before you hear the notification tone.

M3000 Touchphone

To activate Ring Again after hearing a busy signal:

1. Press **Ring Again**.
2. Hang up, or press **Rls**.
3. When you hear the Ring Again tone, lift the handset or select a free **DN**.
4. Touch **Connect**. The number is automatically dialed.

To cancel Ring Again:

— Press **Ring Again** before you hear the notification tone.

M2317 telephone

To activate Ring Again after hearing a busy signal:

1. Press **RINGAGN**.
2. Hang up, or press **Rls**.
3. When you hear the Ring Again tone, lift the handset or select a free **DN**.
4. Press **Call** . The number is automatically dialed.

To cancel Ring Again:

— Press **Call** before you hear the notification tone.

Analog (500/2500 type) telephones

To activate Ring Again after hearing a busy signal:

1 Flash the switchhook or press **LINK**.

2 Dial SPRE+1, or the Flexible Feature Code (FFC) assigned.

3 When you hear the Ring Again tone bursts, lift the handset while you still hear the ringing. The number is automatically dialed.

To cancel Ring Again:

— Before you hear the notification tone, lift the handset and dial SPRE +2, or the FFC assigned, and hang up.

Introduced in X11 Release:	16.67G
Networking:	No

Ring Again on No Answer

The Ring Again No Answer (RANA) feature extends the capabilities of Ring Again for standalone applications, and Network Ring Again for Integrated Services Digital Network (ISDN) applications. The feature allows Ring Again to be applied to a station that does not answer.

This feature applies to Meridian 1 proprietary telephones, as well as analog (500/2500 type) telephones.

Users of Meridian 1 proprietary telephones, upon encountering a station that does not answer, can activate RANA by pressing the Ring Again (RGA) key. When the desired station goes off-hook, to make or receive a call, and then goes on-hook, the station that activated RGA receives a buzz through the telephone's loudspeaker (while the RGA lamp flashes, if that station is idle). The station user can dial the desired station by lifting the handset or pressing a DN key, and then pressing the RGA key.

Users of analog (500/2500 type) telephones, upon encountering a station that does not answer, can activate RANA by performing a recall, and then dialing the Ring Again Activate Flexible Feature Code, or dialing SPRE then the digit 1. After receiving confirmation dial tone, the user goes on-hook to make or receive calls as usual. When the desired station goes off-hook, to make or receive a call, and then goes on-hook, the station that activated RGA receives six ring cycles as a Ring Again notification (if the station is idle). To dial the desired party, the station user has to go off-hook before the six-ring cycle ends. If the desired party goes off-hook while RANA is being applied, Ring Again Busy is activated instead of RANA.

To deactivate RANA from an analog (500/2500 type) telephone, the user goes off-hook and dials the Deactivate Ring Again or the Deactivate Feature Flexible Feature Code, or dials SPRE then the digit 2.

This feature is described more fully in the *International ISDN PRI Feature description and administration* for the ISDN environment.

Operating parameters

Ring Again on No Answer cannot be applied:

— if the dialed DN is a Pilot DN

— to Attendant Consoles

— to a station which has been intercepted to the attendant

— to a station which is queued for an attendant

— to a station which has been recalled to an attendant due to misoperation

— to Automatic Call Distribution (ACD) stations

— to a station with Radio Paging active

— to trunks

Meridian 1 proprietary telephones must be equipped with a Ring Again (RGA) key/lamp combination.

Ring Again on No Answer is applied to the originally dialed DN only.

Feature interactions

Attendant Recall

A set that is recalling the attendant cannot apply Ring Again on No Answer.

Call Forward All Calls
Call Forward No Answer

If an unanswered call is forwarded to another station by any of these features, RANA is applied to the originally dialed station.

Call Forward/Hunt Override Via Flexible Feature Code

Using the Ring Again No Answer feature is possible after using the Call Forward/Hunt Override FFC and encountering an idle set that does not answer. Ring Again No Answer can be placed against the set for which the Call Forward/Hunt Override FFC was used (i.e., the set should be rung by the Ring Again No Answer feature).

Group Hunting

RANA cannot be applied if the DN dialed was a Pilot DN.

Hunting

If RANA has been applied to a station going through a Hunt sequence, Ring Again is applied to that station and not the ringing station.

Intercept Treatment

A telephone that is intercepted to the attendant cannot apply Ring Again on No Answer.

ISDN QSIG/EuroISDN Call Completion

Analog (500/2500 type) sets can have only one Call Completion to Busy Subscriber request at a given time. Meridian 1 proprietary sets can make Ring Again requests based on the number of Ring Again keys programmed on a set.

Multiple Appearance Directory Number

The Ring Again on No Answer feature will only function on Multiple Appearance Directory Numbers that have been assigned to two different sets provided that both users, with the Ring Again on No Answer activated, go off-hook to make a call and then go on-hook. If both users do not go off-hook then the originator will not receive a buzz through the loudspeaker.

Network Intercom

If Ring Again No Answer is activated for a Hot Type I call, it is activated as if the call had been dialed normally.

Phantom Terminal Numbers (TNs)

Although RANA can be applied to a phantom DN, it is not recommended. Because a phantom DN cannot be active or busy, the caller is not notified when the phantom DN's forward DN does not answer.

Queued Calls

RANA cannot be applied by a set which is being queued for the attendant or is in the attendant queue during Night Service.

Recall to Same Attendant

A telephone that is recalling the attendant cannot apply Ring Again on No Answer.

Multiple Appearance Directory Number

The Ring Again on No Answer feature will only function on Multiple Appearance Directory Numbers that have been assigned to two different sets provided that both users, with the Ring Again on No Answer activated, go off-hook to make a call and then go on-hook. If both users do not go off-hook then the originator will not receive a buzz through the loudspeaker.

Telephones - M2317 and M3000

For RANA to function on M2317 and M3000 telephones, the telephones must be configured with a Ring Again (RGA) key. The Ring Again "soft key" will only be displayed when a busy call is encountered and will not be displayed during ring no answer.

Feature packaging

For standalone applications, Ring Again No Answer is part of base X11 system software.

Advanced ISDN Network Services (NTWK) package 148 for network applications.

Feature implementation

LD 15 – Define the Ring Again on No Answer setting.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
...		
OPT		Customer options.
	(RND) RNA	Ring Again on No Answer (denied) allowed.

LD 10 – Define Ring Again for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE 	 500	Type of set. Analog (500/2500 type).
...		
CLS 	 (XRD) XRA	Class of Service options. Ring Again (denied) allowed.

LD 11 – Define Ring Again keys for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
...		
KEY	RGA	Customer options. Ring Again on No Answer (denied) allowed.

Feature operation

Meridian 1 proprietary telephones

Place and Accept Ring Again on No Answer

Action	Response
1. User A calls user B.	User A receives ringback tone.
2. User A presses the Ring Again (RGA) key.	Indicator associated RGA key turns on steadily.
3. User A either goes on-hook or presses the Release (RLS) key.	Indicator associated with RGA key remains on and user A is now free to receive or make other calls.
4. User B, the user against which Ring Again was placed, goes off-hook to make a call, and then back on-hook.	User A is given a short buzz through the loudspeaker and the indicator associated with the RGA key will begin to flash.
5. User A either picks up the handset or presses a DN key.	User A receives dial tone.
6. User A presses the RGA key.	The user against which the Ring Again was placed is rung and the indicator associated with the RGA key is turned off.

Cancel Ring Again No Answer

Action	Response
1. User A presses the RGA key.	The indicator associated with the RGA goes from flashing to off, and ring again is cancelled.

Analog (500/2500 type) telephones

In the following feature operation description, the term recall refers to performing a register recall which may be performed in a number of different ways. Some examples are:

— Flash the switch hook (that is, the equivalent of hanging up the handset and picking it back up. This on-hook, off-hook is performed in a time period that is less than what the system would consider to be a valid disconnect).

— Press the flash or LINK button if equipped.

Place and Accept Ring Again No Answer

Action	Response
1. User A calls user B.	User A receives ringback tone.
2. User A performs a recall.	User B stops ringing and User A receives special dial tone.
Note: User B must be in a ringing state for more than two seconds before recall is allowed.	
3. User A dials either the Ring Again Activate (RGAA) Flexible Feature Code, or the Special Prefix (SPRE) code followed by the digit "1".	User A receives dial tone indicating that the Ring Again was successfully placed.
4. User A goes on-hook.	User A is now free to receive or make other calls.
5. User B, the user against which the Ring Again was placed, goes off-hook to make a call, and then goes back on-hook.	User A is given six cycles of ringing as notification.
6. If User A picks up the handset before all six ringing cycles are complete.	User B is rung.

7. If user A does not pick up the handset before all six ringing cycles are complete.	Ring Again is cancelled.

Cancel Ring Again No Answer

Action	**Response**
1. User A goes off-hook.	User A receives dial tone.
2. User A dials either the Ring Again Deactivate (RGAD) Flexible Feature Code, the Deactivate Feature (DEAF) FFC, or the Special Prefix (SPRE) code followed by the digit "2".	User A receives dial tone indicating that the Ring Again cancellation was successful.

Introduced in X11 Release:	10.10B
Networking:	No

Ring and Hold Lamp Status

The standard lamp-interruption status indication used with the Meridian 1 is 60 impulse per minute (ipm) (flash) for incoming calls and 120 ipm (wink) for held calls on Meridian 1 proprietary telephones, or on terminals emulating Meridian 1 proprietary telephones. This feature, through a Class of Service assigned in LD 11, allows these indicators to be reversed (wink on incoming calls and flash on held calls for all keys that can carry a call, including the group-call key). Data modules with Meridian 1 firmware must use the standard indication of Reverse Lamp Flash Denied Class of Service.

This feature applies to the following key lamps:

— Directory Numbers (DNs)

— Conference

— Transfer

— Voice Call

— Call Waiting

— Dial Intercom Group

— Group Call (For Group Call, a fast blink can be configured to indicate that not all members of a group have answered a group call; a slow flash indicates that a call has been placed on hold by the originator.)

— Automatic Call Distribution (ACD) incalls

— ACD answer agent

— ACD supervisory call, and

— ACD emergency answer.

Operating parameters

This feature cannot be applied to analog (500/2500 type) telephone, M3000 sets, and Attendant Consoles.

This feature is not supported through Attendant Administration.

Feature interactions

Privacy Release

If the Privacy Release feature is activated for multiple-appearance single-call DNs, the blinking rate is based on the Class of Service of each set on which other appearances of the DN occur.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 11 – Create or modify the data blocks for Meridian 1 proprietary telephones.

Prompt	Response	Description
...		
CLS	(RLFD) RLFA	Reversed Lamp Flash (denied) allowed.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	14.47E
Networking:	No

Ringback Tone from Meridian 1 Enhancement

With the current ringback handling, some Public Exchange/Central Office (CO) stations do not send the calling party any ringback tone when calling an analog (500/2500 type) telephone. This enhancement provides a calling-party ringback tone, when a call is placed to a Meridian 1 on a 2 Mbps digital Central Office (CO) trunk.

Operating parameters

There are no feature requirements.

Feature interactions

There are no interactions with other features.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 14 – Configure Meridian 1 Ringback Tone.

Prompt	Response	Description
...		
TYPE	COT	Central Office Trunk data block.
...		
CLS	(CORX) CORP	Central Office Ringback (not) provided by Meridian 1.

Feature operation

Ringback tone is provided until either the call has been answered by an attendant or abandoned by the originator.

Ringing Change Key

This feature allows the user of an M1000 series or digital telephone to change the ringing/non-ringing designation of a Single Call Ringing (SCR) or Multiple Call Ringing (MCR) directory number (DN) located on one of the telephone's key-lamp strips. This is done by using a Ringing Change (RCK) key.

Operating parameters

This feature does not apply to Private Line DNs.

The ringing designation of the Single Call Non-ringing (SCN) and Multiple Call Non-ringing DN keys cannot be changed by using the RCK key.

This feature requires a separate key/lamp configuration.

Feature interactions

Attendant Blocking of Directory Number

When the SACP key (or Signal Source) key is pressed to ring a blocked SCR where the Ring Change feature is activated, an audible ring signal will always be given. This is independent of the Ring Change status.

Directory Number Delayed Ringing

If an SCR/MCR key is toggled from "ringing" to "non-ringing", the Directory Number Delayed Ringing (DNDR) feature will apply to the key. If an SCR/MCR key is toggled again from "non-ringing" to "ringing", the key will be rung immediately and DNDR will no longer apply.

If an SCN/MCN key is toggled from "non-ringing" to "ringing", the DNDR key will ring immediately and DNDR will no longer apply. If an SCN/MCN is toggled again from "ringing" to "non-ringing", the key will not ring immediately and the DNDR feature will apply to the key.

Network Intercom

The ringing/non-ringing mode of an enhanced Hot Type D or of a Hot Type I key is not changeable by using the Ringing Change Key feature.

Feature packaging

International Supplementary Features (SUPP) package 131; and Ringing Change Key (RCK) package 193.

Feature implementation

LD 11 – Define a Ringing Change Key (RCK) for each Meridian 1 proprietary telephone to be equipped with one.

Prompt	Response	Description
...		
KEY	xx RCK y z	Key number, Ringing Change Key, first key lamp strip, second key lamp strip controlled by the key. y = (0)-7 z = 0-(3)-7 Only one RCK key per set is permitted.

Feature operation

Pressing the **RCK** key places the telephone in the Make Set Busy state. Incoming calls to the set receive busy tone, and Multiple Appearance DN calls terminate on another telephone. Pressing an idle **SCR or MCR DN** key indicates the ringing status of the key; a lit key lamp indicates a non-ringing status, and a flashing key lamp indicates a ringing status. Pressing the **SCR or MCR DN** key again changes the ringing status of the key. Pressing the **RCK** again stores the change, and causes the SCR or MCR key lamp to go dark.

During a system initialization a telephone is rendered in the Make Set Busy state. If both the Ringing Change Key and Make Set Busy features are equipped on a telephone, and an initialization occurs during operation of the **RCK** key, the RCK lamp goes dark to inform the user that the changes have not been stored. The MSB lamp is lit to inform the user that the telephone is still in Make Set Busy mode.

Ringing instead of Buzzing on Digital Telephones

The Ringing instead of Buzzing feature, allows a digital telephone to ring when a call is presented as follows:

— when the handset is off hook but the telephone is idle

— when the handset is off hook but the telephone is idle and when the user is busy on another line

Ringing alerts a user in a more obvious way than buzzing (previous operation).

If a call is presented to the telephone, it rings according to the Distinctive Ringing Class of Service (DRG1, DRG2, DRG3, and DRG4), instead of buzzing.

There are two Classes of Service which can be assigned in LD 11:

— RNGI (the set rings when idle but off hook and a call is presented)

— RNGB (the set rings when busy or idle, but off hook and a call is presented.

Operating Parameters

This feature does not affect the features where a buzz is already provided, such as Ring Again or Manual Signaling.

Buzzing is the default configuration.

Ringing features such as Ringing Change Key, Network Distinctive Ringing or Executive Distinctive Ringing, if implemented, affect the way in which the telephone rings.

Any digital telephone can be assigned an RNGI or RNGB Class of Service.

This feature does not affect Attendant Consoles.

If an attempt is made to enter CLS BUZZ, RNGI or RNGB on an analog telephone programmed in LD 11, a service change error message is output.

Feature Interactions

ACD calls

This feature affects calls to an M2216 telephone. Ringing is given to the agent when the CLS is programmed for ringing and the telephone is idle.

Hunting

For telephones with more than one DN, the RNGB Class of Service and Short Hunting programmed calls will ring, not buzz, when the telephone is already busy.

Short Hunting allows calls to hunt to the next higher available key on a proprietary telephone, when a call is already established on a DN key.

Short Buzz for Digital Telephones

The Ringing instead of Buzzing feature takes precedence over the Short Buzz for Digital Telephones feature.

Third Party Applications

Applications which attach to or emulate a digital telephone can be affected by this feature.

Tones, Flexible Incoming

The Ringing instead of Buzzing feature takes precedence over the Tones, Flexible Incoming feature.

Feature Packaging

There are no new software packages for this feature.

Feature Implementation

LD 11 – Configure the Ringing instead of Buzzing feature on digital telephones.

Prompt	**Response**	**Description**
REQ:	CHG	Change.
TYPE:	xxxx	Valid telephone types:
		2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3901, 3902, 3903, 3904, 3905.
TN		Terminal Number
	l s c u	Option 51-81C.
	c u	Option 11C.
CLS	(BUZZ)	Buzz (default).
	RNGI	Ringing applied when telephone is idle but off hook.
	RNGB	Ringing applied when telephone is idle but off hook or busy on the other line.

Feature Operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	All
Networking:	No

Room Status

Room Status allows customers equipped with a Background Terminal (BGD) to store and retrieve data pertinent to the occupancy, readiness, or cleaning status of any guest room or group of guest rooms.

When equipped with the Room Status software, the Meridian 1 system provides the following Room Status information:

— Guest registration and occupancy

OC	(occupied)
VA	(vacant)
CH	(check in)
CH OU	(check out)

— Cleaning status

RE	(cleaning required)
PR	(cleaning in progress)
CL	(room cleaned)
FA	(failed inspection)
PA	(passed inspection)
SK	(cleaning skipped)

— Sale status

NS	(not for sale)
SA	(ready for sale)

— Other status information

CCOS	(Controlled Class of Service)
DND	(Do Not Disturb)
MW	(Message Waiting)
CA	(Category one – 1 to 15)
TL	(telephone check)

Do Not Disturb (DND) has been enhanced for interaction with Room Status on analog (500/2500 type) telephones. A new customer option allows a visual indication of when the analog (500/2500 type) telephone is in the DND mode: The lamp on the telephone lights up.

The Room Status feature provides four methods of accessing the Room Status data:

— Off-hook detection: Hotel and hospital staff generally clean occupied rooms during certain hours of the day. From a Background Terminal (BGD), an option can be entered to set all occupied rooms to "cleaning status request" mode for a predefined time-of-day interval. During this interval, the Meridian 1 system monitors the room telephone's switchhook state to detect a change in the Room Status.

— Dial Access: This method is an enhancement to the off-hook detection method for updating the room cleaning status. This method offers seven cleaning-status options, as compared to the two offered by off-hook detection. Again, you allow or deny the dial access method by using the Background Terminal commands.

— Room Status key: A Room Status key (RMK) can be provided on an SL-1, M1109, or Meridian Modular Telephone. This allows the telephone user to read or alter the status of any room in the system.

— Background Terminal: The Room Status feature is administered from a Background Terminal (BGD) assigned to the customer. BGDs are defined in the configuration record and are connected to the Meridian 1 system through a Serial Data Interface (SDI) port. Devices used as BGDs can be any ASCII serial terminal conforming to EIA RS-232C or CCITT V.24 standards.

Operating parameters

The Room Status key (RMK) is supported only on telephones equipped with a display.

A room telephone is defined with Controlled Class of Service allowed (CCSA). The following telephones are supported as room phones:

— analog (500/2500 type) telephones

— SL-1 and M1309 telephones, and

— Meridian digital telephones.

The M3000, M2317, and ACD telephones are not supported as room phones. Room Status is not supported on telephones with DTA (data terminal allowed) Class of Service. The RMK is not supported on Attendant Consoles.

A room phone is allowed to change the status of its own room.

The Room Status feature is mutually exclusive with the Multiple-Tenant, Centralized Attendant Service (CAS), and Coordinated Dialing Plan (CDP) features.

A message center must be defined for the Do Not Disturb (DND) visual indication function on analog (500/2500 type) telephones. This is mutually exclusive of Integrated Messaging System (IMS) and Meridian Mail Message Centers.

All analog (500/2500 type) telephones that are to use the Do Not Disturb (DND) visual indication must also have an LPA (Lamp Allowed) Class of Service.

Feature interactions

Attendant Administration

Room Status is not supported by Attendant Administration.

Automatic Wake Up

Room Status and Automatic Wake Up both use the Background Terminal (BGD). If the WAKE option is selected for the check-in/check-out operation, the wake-up call for that room is canceled after a check-in or check-out operation.

Automatic Wake Up FFC Delimiter

When a guest has either checked in or out, the room status changes. If an AWU request is still active, it is canceled if it is included as part of the Check In/Out option.

Controlled Class of Service

You can change the access restrictions for room telephones from the BGD or from a telephone equipped with a Room Status key (RMK).

Hot Line

The Room Status feature is incompatible with any telephone for which going off-hook activates Hot Line.

Maid ID

Maid ID is not required but is recommended to track maid performance. The Maid ID must be entered each time the Room Status changes, or it will not be recorded.

Multiple Tenant

Telephones equipped with an RMK can change the Controlled Class of Service (CCOS) of telephones for any tenant in a Customer Group.

Off-Hook Alarm Security

Cleaning changes entered using the Off-Hook Detection Method are mutually exclusive with the Off-Hook Alarm Security (OHAS) feature. OHAS takes precedence over the off-hook detection method of the Room Status feature. If a telephone is defined with the Alarm Security Allowed (ASCA) Class of Service, the off-hook detection method does not work.

Feature packaging

Room Status (RMS) package 100 requires the following:

— Controlled Class of Service (CCOS) package 81, and

— Background Terminal Facility (BGD) package 99.

For lamp status, the requirements are as follows:

— Do Not Disturb, Individual (DNDI) package 9, and

— Message Waiting Center (MWC) package 46.

Feature implementation

Note: This procedure assumes that a BGD has been assigned. Refer to *Background Terminal Facility description* (553-2311-316) for a complete description and list of commands for the Background Terminal.

LD 10 – Add or change Controlled Class of Services (CCOS) for analog (500/2500 type) telephones requiring Room Status updates.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	(CCSD) CCSA	Controlled Class of Service (denied) allowed.

LD 11 – Add or change Room Status key (RMK) for digit display telephones used for Room Status.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, or 2616.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	ADD DDS	Automatic digit display enabled. Digit display enabled.
KEY	xx RMK	Room Status key.

LD 15 – Add or change Customer Data Block to allow (or disallow) visual indication of Do Not Disturb (DND) feature. Offered on the customer level, this applies only to analog (500/2500 type) telephones equipped with a Message Waiting (MW) lamp.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
CUST	0-99 0-31	Customer number. For Option 11C.
- DNDL	YES (NO)	Indicator goes on when DND is active. Indicator does not go on (the default).
...		
TYPE	CCS	Release 21 gate opener.
- CCRS	UNR CUN CTD TLD SRE FRE FR1 FR2	Unrestricted call service. With CCOS active, the restrictions entered apply.

Feature operation

To read the Room Status by using the RMK (display needed):

1 Without lifting the handset, press the **RMK key**.

2 Dial the Directory Number (DN) of the room telephone. The DN is displayed, followed by a dash and a two-digit code.

The first digit indicates occupancy: zero (0) means vacant, one (1) means occupied.

The second digit indicates Room Status:

1 = RE (cleaning required)
2 = PR (cleaning in progress)
3 = CL (cleaned)
4 = PA (passed inspection)
5 = FA (failed inspection)
6 = SK (cleaning skipped), and
7 = NS (not for sale).

To change the Room Status by using the RMK:

1 Without lifting the handset, press the **RMK key**.

2 Dial the Directory Number (DN) of the room telephone.

3 Dial the new room status as follows:

1 = RE (cleaning required)
2 = PR (cleaning in progress)
3 = CL (cleaned)
4 = PA (passed inspection)
5 = FA (failed inspection)
6 = SK (cleaning skipped), or
7 = NS (not for sale).

4 Press the **RMK key**.

To change the Room Status by using Dial Access (from the room telephone):

1 Lift the handset and dial SPRE 86.

2 Dial the room status as shown below:

1 = RE (cleaning required)
2 = PR (cleaning in progress)
3 = CL (cleaned)
4 = PA (passed inspection)
5 = FA (failed inspection)
6 = SK (cleaning skipped), or
7 = NS (not for sale).

3 Dial * and the Maid ID followed by #, if required.

4 Hang up or press **Rls**.

Note: For complete details on the Room Status operation, see *Background Terminal user guide* (P0740427).

Introduced in X11 Release:	14.46E
Networking:	No

Secrecy Enhancement

This feature allows a warning tone to be applied to a three-way connection involving the source, destination and attendant if Warning Tone Allowed (WTA) Class of Service is available on both the source and destination sides. If the warning tone is denied on either the source or destination, these parties are automatically split. This applies to all calls handled by the attendant instead of only incoming network calls and attendant recalls with the original secrecy feature. The warning tone is always applied to a three-way connection.

There will be no connection established through the console with more than two parties, excluding the attendant, unless all parties have WTA Class of Service.

This feature also prevents any intelligible crosstalk on an attendant-held call or if the source (SRC) or destination (DEST) party is excluded.

Operating parameters

A connection is not established through the console if one of the parties, excluding the attendant, has warning tone denied Class of Service.

Feature interactions

AC15 Recall: Timed Reminder Recall

When the attendant answers an AC15 recall, the destination party is excluded from the connection. The attendant is connected to the source party and the excluded destination lamp is lit to show the exclusion of the destination party.

Attendant Break-In with Secrecy

The source and destination parties cannot be joined together on the attendants conference bridge if BKIS is active. This is consistent with the existing Break-In feature.

Attendant Recall

When the attendant answers a recall, the attendant is automatically connected to the destination party and the source party is excluded.

Semi-Automatic Camp-On

Secrecy and Enhanced Secrecy apply to Semi-automatic Camp-On recalls, with splitting taking place when the attendant answers the recall.

Secrecy

All functionalities of the Secrecy feature apply to the Secrecy Enhancement feature.

Slow Answer Recall Enhancement

The Call Waiting Recall and Camp-on Waiting Recall enhancements take precedence over Attendant Recall Splitting (ATS), Secrecy (SYA), Enhanced Secrecy (EHS), and Multiple Party Operations.

Source Included when Attendant Dials

Source Included when Attendant Dials takes precedence over Secrecy and Enhanced Secrecy.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 15 – Modify data for each customer member to be configured:

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
...		
OPT	(SYD) SYA EHS	Secrecy allowed. Enhanced Secrecy allowed. Secrecy denied.

Feature operation

No specific operating procedures are required to use this feature.

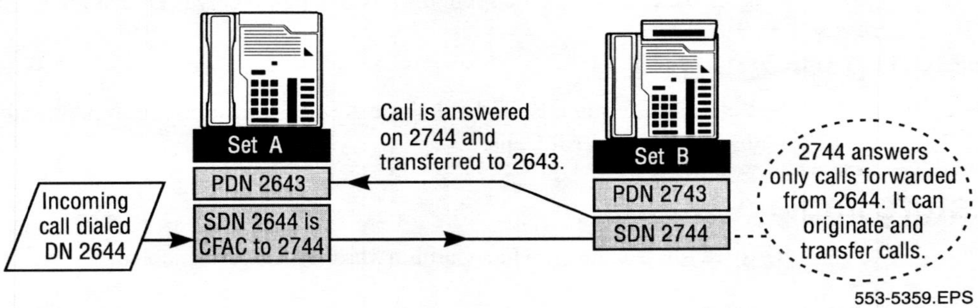
Introduced in X11 Release:	All
Networking:	No

Secretarial Filtering

Secretarial Filtering is an application of Call Forward All Calls. It allows you to forward all calls to a second telephone. The user at the second telephone answers the forwarded calls and can choose to transfer the call back to you.

In the following example, a manager having a secondary DN of 2644 forwards all calls arriving at that DN to a secretary's secondary DN 2744. Any call placed to DN 2644 is forwarded to the secretary at DN 2744. The secretary answers the call, decides that the manager should take the call, and transfers it back to DN 2643 (the prime DN). In this example, the manager receives only the calls originated or transferred by the secretary.

Figure 78
Secretarial Filtering example



Operating parameters

Only the Directory Number (DN) designated as the Call Forward number can originate or transfer calls to the originally dialed DN.

All Single Appearance DNs on the forwarded telephone are forwarded to the target DN.

A Multiple Appearance DN on the forwarded telephone is forwarded only if it is a Prime DN. A Multiple Appearance DN that is not the Prime DN rings at all appearances, including the forwarded telephone.

Feature interactions

Call Forward/Hunt Override Via Flexible Feature Code

The Secretarial Filtering feature is overridden by the Call Forward/Hunt Override Via FFC feature, but there are no changes to the feature itself.

Network Intercom

In a Secretarial filtering scenario, the secretary's BFS lamp also will reflect that the boss' set is busy if the boss is on a Hot Type I call.

Phantom Terminal Numbers (TNs)

If a Phantom TN is call forwarded to an existing telephone, and that telephone is used to call a DN on the Phantom TN, the call receives DCFW treatment.

Feature packaging

Secretarial Filtering is included in base X11 system software. It is provided with Call Forward All Calls.

Feature implementation

This feature is enabled when Call Forward All Calls is enabled.

Feature operation

See the feature operation in the Call Forward All Calls module in this document.

Introduced in X11 Release:	10.10B
Networking:	No

Seizure Acknowledgment

Outgoing Ear and Mouth (E&M) Direct Inward Dialing (DID) or Direct Outward Dialing (DOD) trunks with an immediate start arrangement may require a seizure acknowledgment signal be received after a trunk seizure. This signal is an off-hook message. If the signal is not received within one second of the seizure, the trunk is software busied for three seconds, then dropped. The outgoing call then attempts to seize the next trunk in the sequence to complete the call. If the signal is received, the call is processed normally.

Operating parameters

The Public Exchange/Central Office must be equipped to handle the special signaling requirements associated with the Seizure Acknowledgment feature described above.

The Seizure Acknowledgment feature is not available on 1.5 Mbps digital trunks or Japanese Digital Multiplex Interface (DMI) trunks.

Feature interactions

There are no interactions with other features.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Create or modify data for trunk routes:

Prompt	Response	Description
...		
ACKW	(NO) YES	Seizure acknowledgment signal (is not) is expected after seizure of this DID/DOD trunk.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	23.0x
Networking:	No

Selectable Conferee Display and Disconnect

The Selectable Conferee Display and Disconnect (SCDD) feature expands existing Conference Display functionality and provides Meridian Modular (Aries) set users with the capability to selectively drop any party that has been added to a conference. This feature provides Meridian Modular sets involved in a conference with the following two enhancements:

— Conference Count Display

— Selectable Conferee Disconnect

Note: The Selectable Conferee Display and Disconnect feature applies to Meridian Modular sets equipped with a display screen. The Meridian Modular set must be participating in a conference involving a total of at least three conferees.

Conference Count Display

Previously, only the elapsed time was shown on the display screen of a Meridian Modular set during a conference call. With Conference Count Display, however, the display screen of a Meridian Modular set also shows a count of the number of parties currently active in a conference call. This count includes every conferee involved in the active conference, whether a Meridian Modular set or not. The Conference Count Display is updated whenever a conferee is added to or disconnected from the active conference.

The Conference Count Display is activated at a set level by setting Class of Service to Conferee Display Count Allowed (CDCA). If Class of Service is set to Conferee Display Count Denied (CDCD), the display screen of the Meridian Modular set shows only the elapsed time, as per existing functionality.

The Conference Count Display is composed of three fields which are configured in the Customer Data Block. At least one of these fields must be configured for the Conference Count Display functionality to be in effect. The fields are as follows:

— The Total Conferees Count display field (CNFFIELD) shows the total number of parties involved in a conference (total internal conferees + total external conferees). The default mnemonic for this field on the display screen is "CONF".

— The Total Internal Conferees Count display field (INTFIELD) shows the total number of conferees that are internal to the Meridian 1 system. This includes analog (500/2500 type) sets, Meridian 1 proprietary sets, Attendant Consoles, and service trunks (such as Paging, Music, and Recorded Announcement) within the Meridian 1 system. The default mnemonic for this field on the display screen is "I".

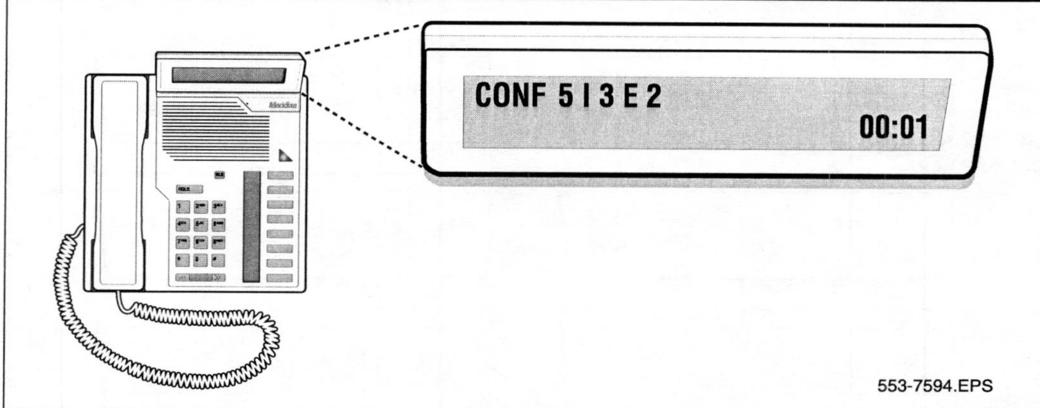
— The Total External Conferees Count display field (EXTFIELD) shows the total number of conferees that are external to the Meridian 1 system. This includes trunks that are connected to the Meridian 1 system that can be configured on internal or external routes. The default mnemonic for this field on the display screen is "E".

The mnemonics for each of the above fields can be modified to accommodate different languages or to save output time. This modification is performed by defining the CNF_NAME, INT_NAME, and EXT_NAME prompts in the Customer Data Block. The mnemonic for each of the three fields can be one to four characters in length.

When modifying the mnemonics for the three display fields, it is recommended that the real time impact be taken into consideration. Since each character, including spaces, is sent to a Meridian Modular set individually, a configuration with the maximum number of characters in each of the field headings (four characters each) affects the refresh time for each of the sets involved in the conference. This is especially important for conferences involving a large number of parties.

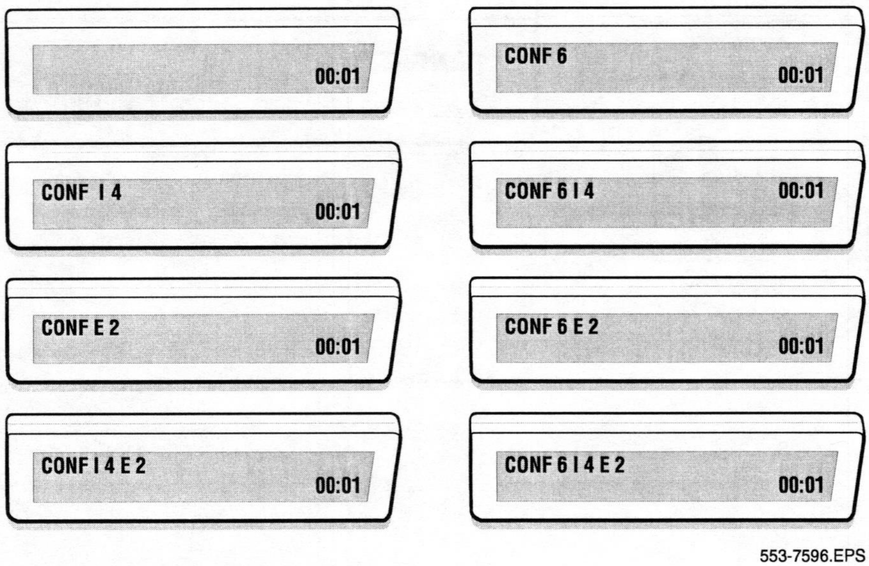
In Figure 79, a Meridian Modular set is involved in a conference consisting of five parties - three internal conferees and two external conferees. The Meridian Modular set has all three Conference Count Display fields (CNFFIELD, INTFIELD, and EXTFIELD) enabled in the Customer Data Block. Also, the Class of Service at a set level is set to Conferee Display Count Allowed (CDCA).

Figure 79
Display Screen of a Meridian Modular set with all three display fields enabled in LD 15



Eight possible Conference Count Display formats can be configured in the Customer Data Block, using a combination of the three Conference Count Display fields. Figure 80 shows the eight possible display formats for Conference Count Display. In this example, a conference has been established with a total of six conferees - four internal parties and two external parties.

Figure 80
Possible display formats of the active Conference Count Display

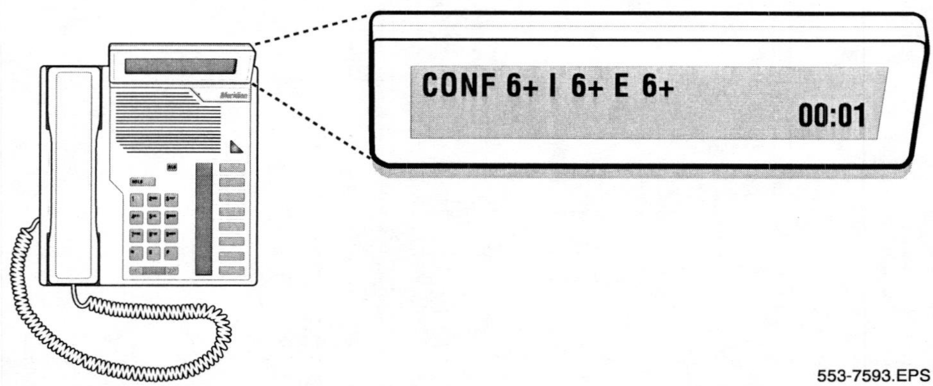


In Figure 80, the Total Conferees Count display field (CNFFIELD) is disabled in the left hand column. CNFFIELD is enabled in the right hand column. The formats in Row 1 have both the Total Internal Conferees Count display field (INTFIELD) and the Total External Conferees Count display field (EXTFIELD) disabled. In Row 2, the INTFIELD is enabled and the EXTFIELD is disabled. In Row 3, the INTFIELD is disabled and the EXTFIELD is enabled. INTFIELD and EXTFIELD are both enabled in Row 4.

Note: The Total Conferees Count display field name (CNF_NAME) is displayed when any of the CNFFIELD, INTFIELD, or EXTFIELD prompts are set to YES in the Customer Data Block.

Each display field on the screen of a Meridian Modular set shows a maximum conferee count of six. If the total number of conferees exceeds six, the Conference Count Display fields show "6+". In Figure 81, a Meridian Modular set is involved in a conference consisting of more than six external parties and more than six internal parties. Therefore, the Total Conferees Count also exceeds six.

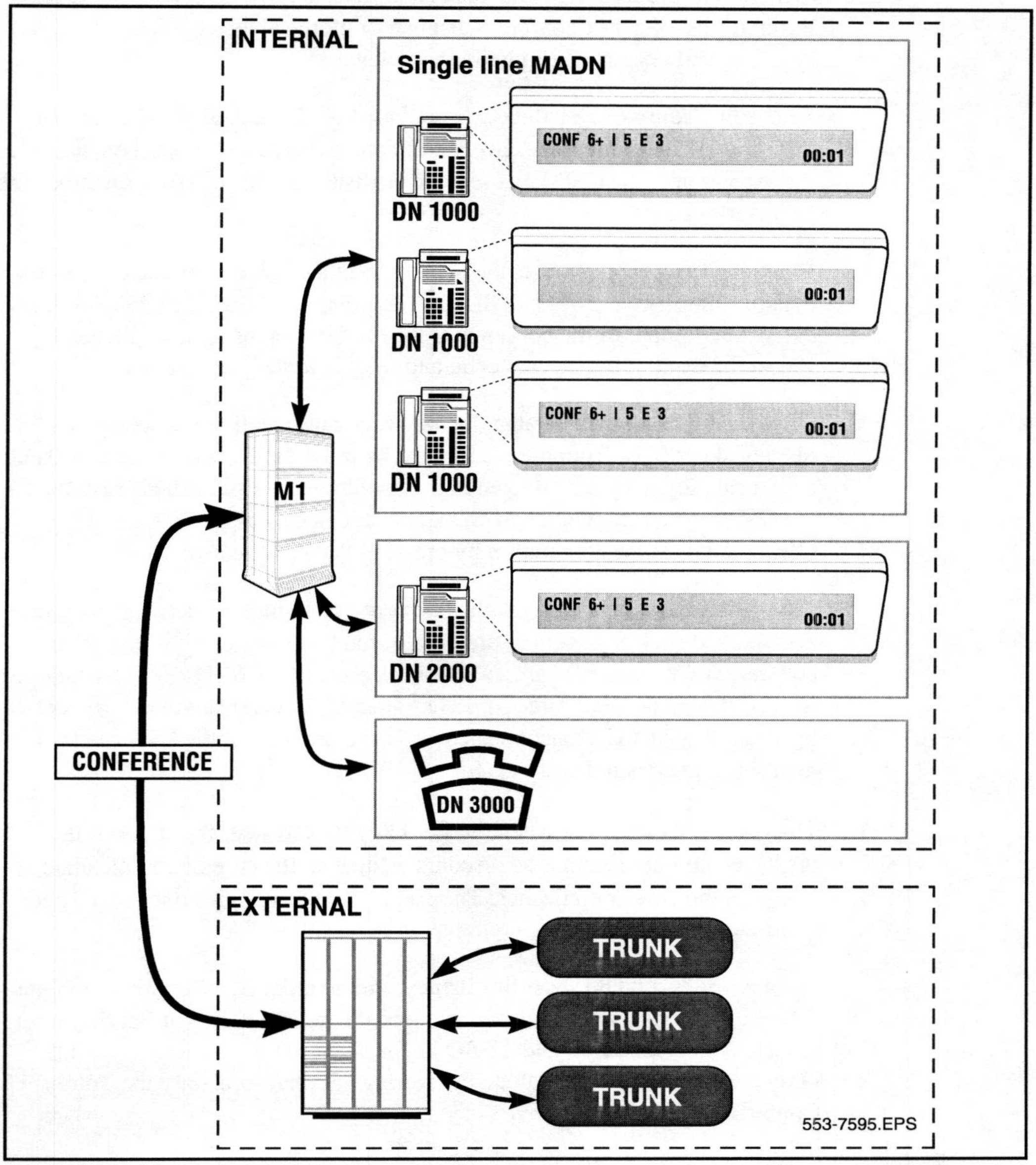
Figure 81
Display Screen of a Meridian Modular set involved in a conference where the total number of internal and external conferees exceeds six



In Figure 82, five internal sets and three external trunks are involved in an active conference. The display screens of the Meridian Modular sets contain Conference Count Display information.

Referring to Figure 82, DN 1000 is a single line Multiple Appearance Directory Number (MADN) on three Meridian Modular sets. All three sets on DN 1000 are involved in the active conference (two of the sets entered the conference via Privacy Override). One of the sets on DN 1000 has Class of Service set to Conferee Display Count Denied (CDCD) in Overlay 11; therefore, its display screen shows only the elapsed time. All other Meridian Modular sets involved in the conference have Class of Service set to Conferee Display Count Allowed (CDCA). DN 2000, a Meridian Modular set, and DN 3000, an analog (500/2500 type) set, are also involved in the active conference. All three display fields are enabled in Overlay 15.

Figure 82
Example of a Conference Scenario involving both internal and external parties



Selectable Conferee Disconnect

With Selectable Conferee Disconnect, a Meridian Modular set user scrolls through a list of active conferees, using a Conferee Selectable Display (CSD) key. The CSD key is configured at a set level.

Selectable Conferee Disconnect is activated when the CSD key is pressed during an active conference. Every conferee involved in the conference, with the exception of the CSD key user, can be displayed one at a time on the CSD key user's screen.

When the CSD key is in use, the display format of each conferee follows the existing simple two party call display. The display shows the name and extension number of the conferee. If the conferee is on a trunk, the display shows the trunk group access code and the trunk member number.

Once the CSD key is activated, the key user can selectively disconnect a displayed conferee from the conference by pressing the active call key. The active call key is the key on which the conference is established. Also once the CSD key is activated, the CSD key user can cancel the Selectable Conferee Disconnect operation by pressing the Release key.

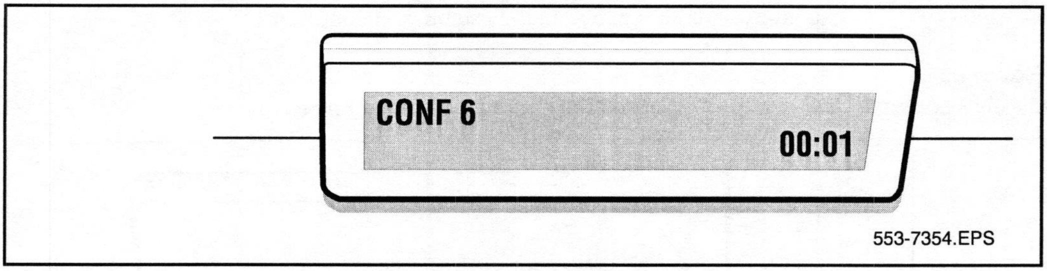
When the CSD key is pressed, the last conferee to join the active conference is displayed first. Subsequent pressing of the CSD key displays the other conferees in no particular order. With each press of the CSD key, the conferee list scrolls in a forward direction. Therefore, if the CSD key user scrolls past the desired party to be disconnected, repeated pressing of the CSD key brings the user to the desired party again.

The CSD key lamp is lit when the CSD key is activated. If, however, the displayed conferee cannot be disconnected, as in the case of an Attendant Console, the Key lamp flashes. The display screen remains unchanged and continues to show the same conferee on the display.

Figures 83, 84, and 85 show the display screen of Set A, a Meridian Modular set, as it displays and disconnects a selected conferee. Class of Service is set to CDCA in Overlay 11 and a CSD key is also configured in Overlay 11. Only the CNFFIELD is enabled in Overlay 15; therefore, only the Total Conferees Count is displayed.

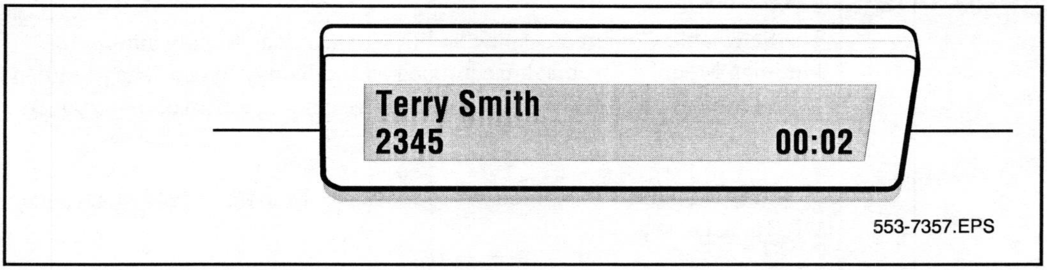
The display screen in Figure 83 shows that there is a total of six conferees involved in an active conference.

Figure 83
Display screen of Set A prior to the CSD key being activated



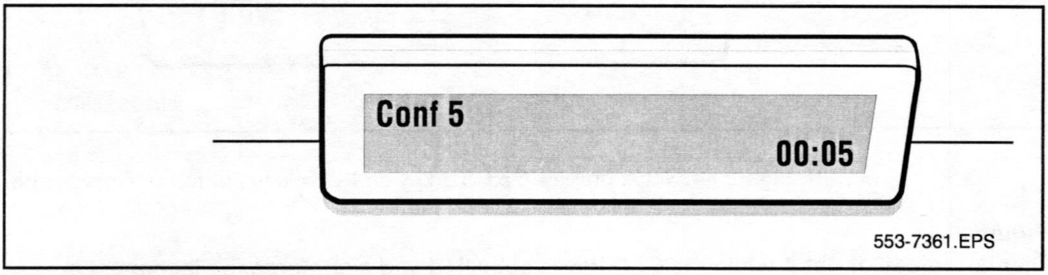
In Figure 84, Set A presses the CSD key and scrolls to conferee, Terry Smith.

Figure 84
Display screen of Set A when the CSD key is activated and a conferee to disconnect is selected



In Figure 85, Set A presses the active call key to disconnect Terry Smith. Set A's display screen is updated to show the new total conferee count status after the conferee is disconnected from the active conference. The Conference Count Displays of the other Meridian Modular sets involved in the conference are also updated. After the disconnection of Terry Smith, a total of five conferees remain active in the established conference.

Figure 85
Display screen of Set A after disconnecting the selected conferee



Operating parameters

The Selectable Conferee Display and Disconnect feature only applies to Meridian Modular sets that are equipped with a display screen. The Meridian Modular set must be participating in an active conference involving a total of at least three conferees.

Meridian Modular sets include M2008, M2016, M2616, M2216ACD1, and M2216ACD2 sets.

When conferees disconnect from an active conference, leaving only two parties in the conference call, the conference is usually converted to a simple two-party call. There are some situations, however, where the two remaining parties are still connected as a conference call. For instance, if either party is an Attendant Console or if both conferees are mixed sets with the same DN, the conference status is maintained.

The Selectable Conferee Display and Disconnect feature is not applicable to two party conferences.

Simple call display for the last two remaining parties in a conference is as per existing operation.

The method that is used to add a conferee to a conference does not affect Selectable Conferee Display and Disconnect. Some of these methods are: 3-party and 6-party Conference, Override, Attendant Barge-In, Attendant Break-In, and Bridging.

The two Selectable Conferee Display and Disconnect sub-features, Conference Count Display and Selectable Conferee Disconnect, have independent functionalities and operations.

Conference Count Display

Conferee Count Display is activated in Overlay 11 by setting Class of Service to Conferee Display Count Allowed (CDCA). At least one of the three Conference Count Display field options must also be enabled in the Customer Data Block.

In order for Class of Service to be set to Conferee Display Count Allowed (CDCA) in Overlay 11, the Automatic Digit Display (ADD) or the Delay Display (DDS) Class of Service must first be set.

A display screen with only the elapsed time showing (existing functionality) can be configured if all three Conference Count Display field options are set to NO in Overlay 15 or if Class of Service is set to Conferee Display Count Denied (CDCD) in Overlay 11.

Selectable Conferee Disconnect

Selectable Conferee Disconnect is activated by defining a Conferee Selectable Display (CSD) key in Overlay 11. Prior to defining the CSD key, however, Automatic Digit Display (ADD) or Delay Display (DDS) Class of Service must be set in Overlay 11.

Only one CSD key can be configured per Meridian Modular set.

The CSD key can only be used during an active conference call.

Each conferee (internal and external) is displayed to the CSD key user following the existing simple two-party call display. No changes are made to the features that supply and/or display the conferee's data. Some of these features are: Call Party Name Display, Calling Party Privacy, Dialed Number Identification Service, Digit Display, Display of Calling Party Denied, and ISDN Calling Line Identification.

For the display of a conferee that is on a trunk, the specific terminating set may not be shown. Therefore, when several trunks are involved in a conference, it recommended that a record be kept of what party joins the conference on what trunk.

When a conferee uses the CSD key, the displays (if any) on the other conferee sets are not changed. Only the CSD key user can see the list of conferees.

After the CSD key is pressed, only the active call key, Release (RLS) key, or CSD key can be used. All other input is ignored.

When the CSD key is activated, if the CSD key user goes on-hook, the key user is disconnected from the call instead of the displayed conferee.

This feature does not support the use of a confirmation tone as indication that a conferee has been disconnected from the active conference.

If the system initializes or sysloads during an active conference, the conference is torn down as per existing functionality. If the CSD key is active when the system initializes or sysloads, then the key operation is canceled.

When the last party to join the conference uses the CSD key, the active conferee list has no particular order. This is because the last conferee to join the conference is the only conferee to be displayed with any priority. In this case, the last party to join the conference is the CSD key user, and the CSD key user is never displayed.

If the last conferee to join the conference is disconnected, then the next scan of the active conferee list has no particular order. The order of inclusion of each conferee is not maintained or stored beyond the last conferee to join the conference.

A conferee can be disconnected from the active conference via the CSD key at any time during the conference call.

When a key or feature key is pressed, the active conference display is replaced. The Conference Count Display is not restored until a conferee is added to or disconnected from the conference, thereby updating the conferee count. If the conference is placed on hold and then restored, the Conference Count Display appears once again.

Feature interactions

Attendant

When the CSD key is activated, the Attendant Console can be displayed as a conferee in the active conference. The CSD key cannot be used to disconnect an Attendant Console from the conference. Only the Attendant Console can release itself from a conference call.

An attempt to disconnect the Attendant Console via the CSD key causes the CSD key lamp to flash. To recover from the flashing CSD key lamp, the key user presses the Release key to cancel the CSD key operation or presses the CSD key again to scroll to the next conferee.

Attendant Barge-In

When an attendant barges into a conference, the conferees are separated. The conferees connected through the trunk that is being verified are placed on the destination (DEST) side and do not include the attendant. The other conferees are conferenced on the Source (SRC) side and include the attendant. However, all parties can communicate with each other.

Once a conference is established on the SRC and/or DEST side, the CSD key is operable. The CSD key, however, cannot be used to disconnect an attendant.

Attendant Break-In

An attendant receives an urgent call and dials the destination DN which is busy. The attendant places the urgent call on hold and then breaks into the active call by using the Break-In key. The destination DN disconnects from the current active call so that the attendant can extend the urgent call.

If the attendant breaks into a simple call, a three-party conference is established including the attendant. Once the conference is established, the new Conference Count Display is not shown. Instead, the displays on the two sets show the attendant information.

If the attendant breaks into a conference call, the attendant is added to the existing conference. Once a conference is established, involving the attendant, the new Conference Count Display is not shown. Instead, the displays of the sets show the attendant information. When the Destination DN disconnects from the active conference, the urgent call is extended to the destination DN. If the remaining parties can form a conference, the Conference Count Display is shown on those sets.

Once a conference is established, the CSD key can be used. However, the urgent call is not shown as a conferee. The attendant is shown as a conferee, but the CSD key cannot be used to disconnect an attendant.

Attendant Administration

Attendant Administration (AA) is modified in order to print the Conferee Selectable Display key when found through Overlay 71. AA cannot be used to configure a CSD key.

Meridian Integrated Conference Bridge

The Selectable Conferee Display and Disconnect feature does not change the functionality of Meridian Integrated Conference Bridge.

Automatic Call Distribution

An Automatic Call Distribution (ACD) agent or supervisor can activate Conference and No Hold Conference. If the ACD set is a Meridian Modular set equipped with a display and a CSD key, then the Selectable Conferee Display and Disconnect feature can be used.

Agent Observe

Selectable Conferee Display and Disconnect does not change the functionality of the ACD Agent Observe feature. While in the observe mode, the ACD supervisor is not part of the conference. Thus, the active conference count does not include the ACD supervisor. The Conference Count Display is not shown on the ACD supervisor's set. When the CSD key is activated, the ACD supervisor is not shown in the active conferees list.

ACD Agent Features

It is recommended that the CSD key not be assigned to agents' sets, as the CSD key can be used to disconnect a supervisor.

Alternate Call Answer

When ACAA = YES in Overlay 23, an agent can put an active Individual DN (IDN) call on hold and then press the In-Calls key to return to the idle agent queue in order to take the next call. If the agent activates Call Join, a conference is established with the agent, the IDN call, and the ACD call.

With Alternate Call Answer, once there is an active conference established with the ACD agent, an IDN call, and the ACD call, the Selectable Conferee Display and Disconnect feature is applicable.

Agent and Supervisor Communication

When the ACD agent is active in a simple call with an ACD caller and wishes to include the ACD supervisor in the call, the ACD agent presses the Answer Supervisor (ASP) key. The supervisor answers by pressing the Agent (AGT) key. In order to finish this operation, the agent presses the ASP key once the supervisor has answered.

When an ACD agent is active in a conference call with an ACD caller, the supervisor cannot be added to the conference via the ASP key.

With Agent and Supervisor Communication, once there is an active conference established, the Selectable Conferee Display and Disconnect feature is applicable.

Emergency Key

The Emergency Key (EMR) feature enables the ACD agent to conference an ACD supervisor and, optionally, a recording device for customer-defined emergencies or sensitive situations.

When the EMR key is activated, the recording trunk is not considered a member of the conference. When the CSD key is activated, the recording trunk is not included in the active conferees list. The total number of conferees on the Conference Count Display does not include the recording trunk.

ACD Display Enhancement

With the ACD Display Enhancement, the Not Ready (NRD) key cannot be pressed when using the Conference key. When a conference is established and the NRD key is pressed, the conference call is disconnected. In this case, the NRD key lamp is lit, and the 'NOT READY' screen is displayed.

When the CSD key is active, the NRD key cannot be used. Pressing the NRD key is ineffective.

ACD In-Calls Key

When a conference is established on the ACD In-Calls key, the In-Calls key is used to drop a desired conferee when the CSD key is activated. The Position Identification (POS ID) of each ACD set involved in the conference is displayed when the CSD key user scrolls through the active conferees list.

Application Module Base

The Selectable Conferee Display and Disconnect feature uses the existing messaging to disconnect a conferee. The messaging to disconnect a conferee is the same as though the conferee has gone on-hook or has pressed the Release (RLS) key to disconnect themselves from the conference.

Automatic Hold

The Selectable Conferee Display and Disconnect feature does not change the functionality of the Automatic Hold feature. Once a conference is established on the active DN key, the Selectable Conferee Display and Disconnect feature is applicable.

Basic Rate Interface

The Selectable Conferee Display and Disconnect feature is not supported on BRI sets. However, if a conferee involved in the active conference is on a BRI set, its information is shown when the CSD key is activated.

Bridging

The Selectable Conferee Display and Disconnect feature does not change the functionality of the Bridging feature.

With the Bridging feature, the same DN can appear on up to eight single-line sets. Any appearance of the MADN can enter a call by going off-hook. When a conference with three parties is created through Bridging, there are only two active DNs in the conference call. As long as there are only two different DNs in the bridged conference call, the displays on the sets show the information of the other DN involved in the call, not the Conference Count Display information. In this case, however, the CSD key can be used, as more than three conferees are active in the conference call.

Once there are more than two different DNs in the conference call, the Conference Count Display shows the count of the conferees. Once a conference is established, the CSD key is applicable.

Conference

The Selectable Conferee Display and Disconnect feature does not change the functionality of Conference, except for the new active conference display. Conference calls can include calls on the following key types: Single Call Arrangement DN (SCN, SCR), Multiple Call Arrangement DN (MCN, MCR), ACD In-Calls (ACD DN), Private Line Ringing and Non-ringing (PLN, PLR), Hotline (HOT), Call Waiting (CWT), Voice Call (VCC) and Dial Intercom (DIG).

Conference Control

The Selectable Conferee Display and Disconnect feature does not change the functionality of the Conference Control feature.

Digitone Receiver

The Selectable Conferee Display and Disconnect feature does not treat the Digitone Receiver (DTR) as a conferee when it appears on the conference loop since it appears only temporarily to provide the tone service.

Display Key

While in a conference call, the Display (DSP) key can be used to obtain information. However, the Display key is blocked when the CSD key is active.

DNIS Across Call Modifications

When a CSD key user scrolls through the list of conferees during a DNIS call, the DNIS information is displayed.

End-to-End Signaling

The Selectable Conferee Display and Disconnect feature does not block End-to-End Signaling (EES) or dialing digits while the CSD key is active.

Group Call

The Selectable Conferee Display and Disconnect feature is only applicable to the originator of a Group Call involving three or more active parties. The active conference display is not shown until a redisplay of the Group Call originator's screen is needed.

Hold

With the Selectable Conferee Display and Disconnect feature, when a Meridian Modular set equipped with display is involved in a conference, its display shows the Conference Count Display. If a Meridian Modular set puts the conference on hold by pressing the Hold key, the active DN key lamp flashes, and the display is cleared during the held operation. The Conference Count Display is restored upon completion of the held operation. The active DN key is pressed to restore the held conference call.

Meridian Link

The Selectable Conferee Display and Disconnect feature uses existing messages sent over the Meridian Link in order to provide the Conference Count Display and the Selectable Conferee Disconnect functionality.

No Hold Conference

The Selectable Conferee Display and Disconnect feature does not change the No Hold Conference (NHC) functionality. The Selectable Conferee Display and Disconnect feature is applicable to conferences created by No Hold Conference.

Override

The Selectable Conferee Display and Disconnect feature does not affect the operation of the Override (OVR) feature. The Conference Count Display is not shown for an Override conference, as the Override display is shown instead. The CSD key, however, can be used to disconnect conferees in an Override conference.

Priority Override

The Selectable Conferee Display and Disconnect feature does not affect the operation of the Priority Override (POVR) feature. The Conference Count Display is not shown for a POVR conference, as the Priority Override display is shown instead. The CSD key can, however, be used to disconnect conferees involved in a POVR conference.

Privacy

The Selectable Conferee Display and Disconnect feature does not affect the operation of the Privacy feature. With Privacy enabled, only one appearance of a single line MADN can participate in a conference call. This appearance is included in the conferee counts.

Privacy Override

The Selectable Conferee Display and Disconnect feature does not change the operation of the Privacy Override (POA) feature.

A Meridian 1 proprietary set with Privacy Override Allowed (POA) Class of Service can bridge into an established call on a single line MADN. When a conference with three parties is created through Privacy Override, there are only two active DNs in the conference call. As long as there are only two different DNs in the POA bridged conference call, the displays on the sets show the information of the other DN involved in the call, not the Conference Count Display information. In this case, however, the CSD key can be used, as more than three conferees are active in the conference call.

Once there are more than two different DNs involved in the active conference call, the Conference Count Display shows the count of conferees. The conferees that are added to the conference through POA are included in the Conference Count Display totals. Once a conference is established, the CSD key is applicable.

Tone and Digit Switch

The Selectable Conferee Display and Disconnect feature does not treat the Tone and Digit Switch (TDS) as a conferee when it appears on the conference loop, as it appears only temporarily to provide tone service.

Feature packaging

Selectable Conferee Display and Disconnect is included in base X11 System Software.

Feature implementation

LD 15 – Enable and modify the Conference Count Display Format.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options.
CUST	xx	Customer number.
...		
CONF_DSP	YES	Change Conference Count Display configurations. NO = Do not change Conference Count Display configurations (default). To prompt for further conference display options, CONF_DSP must be set to YES.
- CNFFIELD	(NO) YES	Total Conferees Count display field (disabled) enabled.
- CNF_NAME	(CONF) aaaa	Total Conferees Count display field name. Enter 1-4 alphanumeric characters to replace the existing name. The Total Conferees Count display field name is displayed when any of the CNFFIELD, INTFIELD, or EXTFIELD prompts are set to YES.
- INTFIELD	(NO) YES	Total Internal Conferees Count display field (disabled) enabled.
--INT_NAME	(I) aaaa	Total Internal Conferees Count display field name. Enter 1 to 4 alphanumeric characters to replace the existing name.
- EXTFIELD	(NO) YES	Total External Conferees Count display field (disabled) enabled.
--EXT_NAME	(E) aaaa	Total External Conferees Count display field name. Enter 1 to 4 alphanumeric characters to replace existing name.

LD 11 – Set the Conferee Display Count Allowed (CDCA) Class of Service for Meridian Modular sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Telephone type where xxxx is: 2008, 2016, 2216, or 2616.
TN	l s c u c u	Terminal Number. For Option 11C.
...		
CLS	ADD DDS	Automatic Digit Display. Delay Display. With CLS = DDS, the display is activated after the call is answered. CLS must be set to either ADD or DDS prior to setting CLS = CDCA or CDCD.
	(CDCA)	Conferee Display Count Allowed (default) CDCD = Conferee Display Count Denied. CDCD option sets a blank display screen during a conference call.

LD 11 – Add or change a Conferee Selectable Disconnect (CSD) key for Meridian Modular sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Telephone type where xxxx is: 2008, 2016, 2216, and 2616.
TN	l s c u c u	Terminal Number. For Option 11C.
...		

CLS	ADD DDS	Automatic Digit Display. Delay Display. With CLS = DDS, the display is activated after the call is answered. CLS must be set to either ADD or DDS prior to configuring a CSD key.
KEY	xx CSD	Conferee Selectable Display key. To remove the CSD key, set the KEY prompt to xx NUL, thereby disabling Selectable Conferee Disconnect.

Feature operation

Viewing the list of active conferees

To view the list of active conferees:

1. Press the Conferee Selectable Display (CSD) key to view the list of active conferees. Continue to press the CSD key to view each conferee. The CSD key lamp is lit. The displays on the other Meridian Modular sets involved in the conference are not changed.

2. Press the Release key to cancel the Selectable Conferee Disconnect operation. None of the conferees are disconnected. The CSD key lamp is dark. The Conference Count Display returns if it is enabled. The original conference call remains active throughout this operation.

Disconnecting one conferee

To disconnect a conferee using the CSD key:

1. Press the CSD key repeatedly until the conferee that is to be disconnected is displayed on the screen. The CSD key lamp is lit. The displays on other Meridian Modular sets are not changed.

2. Press the active call key (the key on which the active conference is established). The displayed conferee is disconnected. The CSD key lamp is dark. The Conference Count Display returns, if enabled, showing the revised total count of conferees. The original conference call remains active throughout this operation.

Disconnecting more than one conferee

In order to disconnect more than one conferee, follow the steps for disconnecting one conferee. Each conferee must be disconnected separately.

> ***Note:*** When two CSD key users wish to drop different conferees (but not each other), each CSD key user can initiate the Selectable Conferee Disconnect operation and disconnect the selected conferee. If enabled, the Conference Count Displays on the Meridian Modular sets are revised once each Selectable Conferee Disconnect operation has concluded successfully.

Disconnecting the same conferee

Two Meridian Modular sets (Set A and Set B), both equipped with a CSD key, wish to disconnect the same conferee. The Set that presses the active call key first is successful in disconnecting the conferee. If Set A is the first set to press the active call key, its Conference Count Display is updated with the revised total count of conferees. The Conference Count Display of all other Meridian Modular sets, with the exception of Set B, are also updated. Set B's Conference Count Display is updated when it presses the active call key or when it presses the Release key to end the operation.

Verifying that a conferee has been disconnected

To verify that a conferee has been disconnected:

— View the list of conferees using the CSD key, and note whether or not the disconnected conferee is still listed.

— Check that the CSD key lamp is dark. This indicates that the Conferee Selectable Disconnect operation is complete.

— Check that the total count of conferees on the Conference Count Display has been revised on the display screen.

— If the conferee is disconnected and only two parties remain, a simple call situation is established. Therefore, the displays are updated accordingly.

Canceling the Selectable Conferee Disconnect operation

To cancel Selectable Conferee Disconnect operation at any time, press the Release key when the Conferee Selectable Disconnect operation is in progress. When the Release key is pressed, none of the conferees are disconnected, the CSD key lamp is dark, and the Conference Count Display returns (if enabled). The original conference call remains active throughout this operation.

Disconnecting from an active conference

To disconnect yourself from an active conference, press the Release key or go on-hook. In this case the original conference call remains active, as long as a supervised conference situation remains.

Introduced in X11 Release:	10.10B
Networking:	No

Selectable Directory Number Size

The Selectable Directory Number Size feature allows a user to define the number of digits that must be received on a Direct Inward Dialing (DID) route before the end of dialing (EOD) is reached. If the required number of digits is not received when the EOD timer expires, a TRK137 message is sent to print and the trunk is locked out.

The DN size can be specified from one to seven digits, or as zero which will not consider the number of digits dialed in the sequence.

Operating parameters

The Public Exchange/Central Office must be equipped to handle the special signaling requirements associated with the Seizure Acknowledgment feature described above.

The Seizure Acknowledgment feature is not available on 1.5 Mbps digital trunks or Japanese Digital Multiplex Interface (DMI) trunks.

Feature interactions

There are no interactions with other features.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Create or modify data for trunk routes.

Prompt	Response	Description
...		
DNSZ	(0)-7	Number of digits expected on DID routes; 0 indicates no fixed number.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	15.58F
Networking:	No

Semi-Automatic Camp-On

This feature allows a Camp-On call to recall to the attendant instead of ringing the called party when the called party becomes available. The called party can originate calls but cannot receive any other calls. Other incoming calls to this DN will receive a busy indication. If the called party originates another call when the attendant attempts to present the Camp-On call, the attendant receives busy tone and can initiate Camp On again or release the call.

When an attendant extends a call to a desired party that is busy, the attendant can activate Semi-automatic Camp-On by pressing the Semi-automatic Camp-On (SACP) key. This causes the call to be camped-on to the desired party, and recalled to the attendant when the desired party becomes idle, rather than rung through to the desired party.

Recall to Same Attendant must be allowed, otherwise the recall is routed to the first available attendant. The attendant display shows the calling-party DN and the party to which the call is camped-on. If the attendant, or all attendants in a multiple-console environment, are busy then the recall is placed in the attendant queue.

Meanwhile, incoming calls to the desired party receive busy treatment. The desired party, however, is still able to make calls. After receiving the recall, the attendant can ring the desired party by pressing the SACP key. The attendant may release the call while it is ringing, or hold the call until it is answered. If the desired party has made another call while the attendant tries to present the recall, the attendant may Camp-On the recall to the desired party by pressing the SACP key.

Operating parameters

The same operating parameters apply as for Camp-On.

Semi-automatic Camp-on is mutually exclusive with the Call Waiting feature. Thus, Attendant Consoles configured with Semi-automatic Camp-on will not work if Call Waiting has been defined.

Semi-automatic Camp-On can be configured for individual or all Camp-On occurrences.

Semi-automatic Camp-On is not available with Network Attendant Service. If the attendant tries to apply Semi-automatic Camp-On to a station at a remote node, the SACP lamp flashes to indicate that Semi-automatic Camp-On is not allowed. The attendant has to press the SACP key again to deactivate the feature, and be allowed to activate it under normal operation.

Semi-automatic Camp-On is not supported during Night Service or Enhanced Night Service. Calls that were camped-on by Semi-automatic Camp-On during normal hours ring through to the desired party, when idle, and do not recall to the attendant.

Feature interactions

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature uses the SACP key to activate a blocking attempt, but the Attendant Blocking of DN feature is only valid on the source side of the Attendant Console. The Semi-automatic Camp-on feature is only valid on the destination side of the Attendant Console.

To have the Attendant Blocking of DN feature available and not the Semi-automatic Camp-on feature, a new response to the SACP prompt has been introduced in LD 15. Prompt SACP = NO means the Semi-automatic Camp-on feature is not available even if the SACP package is equipped and an SACP key exists on the Attendant Console. To have the Semi-automatic Camp-on feature available the SACP prompt must be answered with SNGL or ALL which have the same meanings as before.

Attendant Break-In

The attendant can Break-In to an established call and apply Semi-automatic Camp-On to the desired party. The attendant may press the SACP key before or after the Break-In.

Call Forward/Hunt Override Via Flexible Feature Code

Semi-Automatic Camp-On can be used even if the Call Forward/Hunt Override Via FFC feature is activated. When encountering a busy set, it is possible to activate SACP, if it is applicable.

Incoming calls during recall

During Semi-automatic Camp-On, when the desired party becomes idle and the camp-on is recalled to the attendant, the desired party appears busy to incoming calls. The DN of the desired party is displayed as busy on the Busy Lamp/Enhanced Busy Lamp display.

Periodic Camp-On Tone

Periodic Camp-On Tone stops when the camped-on call is recalled to the attendant.

Secrecy Enhancement

Secrecy Enhancement applies to Semi-automatic Camp-On recalls, with splitting taking place when the attendant answers the recall.

Source Included when Attendant Dials

The source remains included while the attendant dials the destination.

Feature packaging

Semi-automatic Camp-On (SACP) is package 181.

Feature implementation

LD 15 – Configure Semi-automatic Camp-On.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	CDB ATT	Customer Data Block. Release 21 gate opener.
...		
RTSA	RSAA	Recall To Same Attendant Allowed.
SACP	 (NO) SNGL ALL	Semi-automatic Camp-On. Semi-automatic Camp-On not allowed. Enable Semi-automatic Camp-On on a per-call basis. Enable Semi-automatic Camp-On for all occurrences. SACP keys must be defined on all Attendant Consoles which are to make use of the feature.

LD 12 – Configure an SACP key on the Attendant Console.

Prompt	Response	Description
...		
KEY	xx SACP	Key number, Semi-automatic Camp-On.

Feature operation

When an attendant extends a call to a desired party who is busy, the attendant can activate Semi-automatic Camp-On as follows:

1. Press the SACP on the Attendant Console.
   The call is camped on the desired party.

   The display on the Attendant Console shows the calling party's DN, and the party to which the call is camped on (the desired party).

2. The desired party becomes idle.
   The call is recalled to the attendant.

3. To ring the desired party after receiving the recall, press the SACP on the Attendant Console again.

Recall timing on Camp-On calls

When any station extends an external call, recall timing will be initiated if the call is camped on to a busy station.

The recall timing will start from the moment that the extending station "releases" the call. The value of the recall timer is set by the prompt RTIM in the Customer Data Block (LD 15).

At the recall, the camped on call will be routed to the attendant. If the attendant is in Night Service, Night treatment is given; if NAS routing is active, the call will be routed according to the NAS configuration.

Standalone case

When the recall to the attendant occurs, the Camp-On is canceled. If the attendant is busy during the recall, the recall will be queued.

Network case

When the recall occurs and the attendant has answered the recall, the call will still be camped on to the desired party. If during the recall the attendant is busy, the recall will be queued.

Introduced in X11 Release:	15.58F
Networking:	No

Series Call

The Series Call feature causes a source call (either an attendant-answered incoming call, or an attendant-originated trunk call), that has been extended to an internal destination party, to be recalled to the attendant when the destination party hangs up. The attendant can then extend the source call to another destination party. This feature enables a caller to talk to more than one party without having to disconnect and call again (Recall to Same Attendant must be allowed, otherwise the recall is routed to the first available attendant). This process can be repeated for as many destinations as requested by the caller.

A Series Call is canceled if one of the following occurs:

— the attendant presses the Series Call (SECL) key while the associated lamp is lit

— the attendant extends the source to a trunk while the SECL lamp is lit

— the attendant enters Night Service after extending the call and prior to receiving the recall

— the destination is call forwarded to a trunk, or

— the source disconnects.

Operating parameters

This feature only applies when the destination party is internal. If the attendant dials a DN that is not internal, the SECL key will flash to indicate that the feature cannot be invoked.

Feature interactions

Attendant Position Busy

If the attendant activates Position Busy while a Series Call is active, the recall occurs to the next available attendant.

Call Detail Recording

With Call Detail Recording, a start record is generated when a source Periodic Pulse Metering call is answered and marked as a Series Call by the attendant, and an end record is generated when the attendant releases the call. No intermediate records are generated.

Night Service

If the attendant extends a Series Call and goes into Night Service before it recalls to the attendant, the call recalls to the night DN and Series Call treatment is canceled.

Timed Reminder Recall

With Timed Reminder Recall, if the attendant extends a Series Call during Camp-on, Call Waiting, or ringing, the SECL lamp goes dark.

Feature packaging

Series Call (SECL) is package 191.

Feature implementation

LD 12 – Series Call keys must be defined for each Attendant Console to make use of the function.

Prompt	Response	Description
...		
KEY	xx SECL	Key number, Series Call.

LD 15 – Recall to the first available attendant.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT	Change Attendant Console options.
CUST	0-99	Customer number.
...		
- RTSA	(RSAD), RSAA	Recall (Denied) Allowed to Same Attendant.

Feature operation

The attendant designates the source call as a Series Call by pressing the Series Call (**SECL**) key. The **SECL** key may be pressed by the attendant while dialing, talking to the destination party, or while a call is ringing. The associated key lamp remains lit until the Series Call is canceled. If the attendant tries to extend a call to an external station, the SECL lamp flashes. The attendant has to press the **SECL** key to cancel the Series Call, and extend the call as a standard call extension.

Introduced in X11 Release:
Networking:

Set-Based Administration

Previously, Set-Based Administration was a feature available on Meridian 1 Option 11 systems that simplified system installation and administration by enabling a set to be used to perform several administrative and maintenance procedures. With X11 Release 21 and the Set-Based Administration Enhancements feature (ADMINSET package 256), Set-Based Administration is now available for Options 21E through 81C. In addition, enhancements are provided to the existing capabilities on the Option 11.

For further information on Set-Based Administration, please refer to the *Set-Based Administration* NTP 553-3001-303.

Introduced in X11 Release:	21
Networking:	No

Set-Based Administration Enhancements

Previously, Set-Based Administration was a feature available on Meridian 1 Option 11 systems that simplified system installation and administration by enabling a set to be used to perform several administrative and maintenance procedures. With the Set-Based Administration Enhancements feature, Set-Based Administration is now available for all system types. In addition, enhancements are provided to the existing capabilities on the Option 11.

To further enhance Set-Based Administration, three levels of set-based data administration access are available, having the following capabilities:

— Administrator Access allows a system administrator to make changes to any supported telephones within the same customer location. The system administrator can perform any of the following tasks through an administration/maintenance set (M2008, M2016, M2216, M2616 with display):

- Change the data associated with specific set-related features (that is, Hunting, External Hunting, Call Forward No Answer, External Call Forward No Answer, Call Forward, Busy Forward Status, Voice Call, Dial Intercom Group, Group Call, Ringing Number Pickup Group, Speed Call, System Speed Call, and Hot Line)
- Add or change the Calling Party Name Display (CPND) names associated with existing DNs
- Change system date and time
- Change toll restrictions of any set, and
- Determine Directory Number-Terminal Number correspondence.

— Installer Access allows an installer to perform any of the following tasks to a set from which the installer is logged in:

- Change the data associated with specific set-related features
- Add or change the Calling Party Name Display names associated with the DN on that set
- Change system data and time, and
- Change toll restriction for that set.

— User Installation allows a user to add or change the user's own CPND when logging in through the user's own set.

Administrator and Installer Access are invoked by dialing the Administrator or Installer Flexible Feature Code (FFC) followed by the Administrator or Installer password. The passwords are defined on a system basis. User Access is activated by dialing the Set-Based Administration User FFC followed by the Station Control Password of the user's set.

The multi-language capability of this feature supports all languages currently supported on the Option 11. These languages are English, German, Spanish, Swedish, Canadian and Parisian French, Dutch, Italian, Danish, Portuguese, and Norwegian.

For the Option 11 the functionalities that have been offered by Set-Based Administration prior to Release 21 are now grouped under the following two tasks on the main menu, under administration access:

— Administration: provides a grouping of trunk-related options.

— Installation options: provides the same functions as before; however, it is moved to a new location on the main menu.

Since the above two capabilities are only available to Option 11, they will not be displayed on the main menu for other system types.

For more information about the Set-based Administration Enhancements feature, please see *Set-Based Administration* 553-3001-303.

Introduced in X11 Release:	12
Networking:	No

Short Buzz for Digital Telephones

When a call is presented to a digital telephone that is off-hook, a buzz tone is given. The duration of this secondary buzz is shortened from two seconds to an average of 0.8 seconds, with a minimum length of 0.5 seconds and a maximum length of one second.

Operating parameters

Short Buzz for digital telephones does not apply to SL-1 telephones.

Short Buzz for digital sets does not change the buzz tone given to Automatic Call Distribution (ACD) telephones on the In-calls key.

Feature interactions

Directory Number Delayed Ringing

If a set is defined with Directory Number Delayed Ringing (DNDR) delay and there is an incoming call to another SCN/MCN DN key on the same set, buzzing (or short buzzing) is applied after the DNDR delay timer expires.

Group Call

The special three-second buzz for Group Call is not affected by this feature.

Feature packaging

Short Buzz for Digital Telephones is included in base X11 system software.

Feature implementation

No change to existing configuration is required to implement this feature.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	15.58F
Networking:	No

Single-digit Access to Hotel Services

In hospitality applications, it is desirable for room phones to have single-digit access to hotel services and a multiple-digit access to room phones.

The Single-digit Access to Hotel Services feature allows a customer to define a pause timer, called a second-digit timer, between the first and second dialed digits, and allows two speed-call entries to be defined for a station group. The first speed-call entry is used for normal pretranslation. The second speed-call list is used when the second digit timer times out (that is, when time out occurs after the first digit is dialed, with the first digit in the first speed-call list being translated).

Operating parameters

There are no feature requirements.

Feature interactions

There are no interactions with other features.

Feature packaging

Single-digit Access to Hotel Services requires International Supplementary Features (SUPP) package 131.

Dependency:

— Pretranslation (PXLT) package 92

Feature implementation

LD 15 – The Single Digit Access to Hotel Services feature must be enabled on a customer basis.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
...		
- OPT	(SDDE) SDAL	(Deny) allow Single Digit Access.

LD 18 – Translation tables used by this feature must be defined.

Prompt	Response	Description
...		
XLAT	xxx yyyy	Translate. Calling group number to translation Speed Call list number correlation. Format if SUPP package 131 is not equipped, where: xxx = Pretranslation group number, 0-254. Group 0 is used for trunks. Group 1 is used for Attendant Consoles. Groups 2-254 can be used for other calling groups. yyyy = List number to be used for Pretranslation, 0-8191 (8191 is used to remove the group from pretranslation). Format if SUPP package 131 is equipped, where: xxx = Pretranslation group number. Group 0 is used for trunks. Group 1 is used for Attendant Consoles. Group 2-254 can be used for other calling groups.
- SDA	0-8190	Single-digit Access Speed Call List number.

Feature operation

In the example that follows, if a room guest dials the digit 7, the guest's call is immediately terminated at DN 4300, the front desk. If the guest had dialed the digit 2, then after the second digit timer times out, the guest's call is terminated at DN 4002, laundry. If the guest enters three more digits (*xxx*) before the second digit time-out, the appropriate room number (2*xxx*) is rung.

Speed Call CodeDN Designation

0 Operator (00)
0101-393900-99 room service, floors 1-39
1 Room Service (4001)
2 Laundry (4002)
3 Concierge (4100)
4 Restaurant (4101)
5 Health Club (4200)
6 Maid (4201)
7 Front Desk (4300)
8 Toll Calls (88)
9 Local Calls (99)

First Entry Speed Call List (for normal pretranslation)

First Dialed DigitAction

1 Pass as 1
2 Pass as 2
3 Pass as 3
4 4101
5 4200
6 4201
7 4300
8 88
9 99
0 Pass as 0

Second Entry Speed Call List (for pretranslation after time out)

First Dialed Digit	Action
1	4000
2	4002
3	4100
4	N/A
5	N/A
6	N/A
7	N/A
8	N/A
9	N/A
0	0017

Introduced in X11 Release:	15.59F
Networking:	No

Slow Answer Recall Enhancement

This enhancement to the Slow Answer Recall feature changes how the recall is treated once presented to the Attendant Console. This enhancement applies to Integrated Services Digital Network (ISDN) and standalone environments.

If an incoming call extended by the attendant to a set is not answered after a preprogrammed time period, it is recalled to the Attendant Console. The call type may be indicated by an Incoming Call Indicator (ICI) key programmed to flash for recalls. The target set will continue to ring after the call is presented to the attendant. The target set can answer the call before the attendant does, in which case the call is cleared from the Attendant Console and the incoming call and target set will be connected.

If the attendant answers the recall before the target set, a speech connection is established between the calling party on the source (SRC) side of the console. The target set continues to ring while still being connected to the destination (DEST) side of the console. This feature only affects the operation after the attendant has answered the recall.

In a ISDN environment, the feature works in a similar way regardless of the location of the called party (on the same node as the attendant or on a remote node), and if Network Attendant Service (NAS) routing is involved in the call or not.

Call Waiting Recalls and Camp-on Recalls

This enhancement adds Call Waiting Recall and Camp-on Recall functionality to Slow Answer Recall. This enhancement applies within standalone and networking environments.

Call Waiting Recall

Within a standalone environment, if an incoming call extended by the attendant or a set (equipped with the Multi-Party Operations feature) to a busy station (equipped with Call Waiting) is not answered within a customer-defined period of time, it is recalled to the attendant. The recall is presented to the attendant or placed in the attendant queue.

Camp-on Recall

Within a standalone environment, an incoming call is extended by the attendant or a set (equipped with the Station Camp-on feature) to a busy station that is not equipped with Call Waiting. The attendant or set camps on the call to the target set. If the call is not answered within a customer-defined period of time, it is recalled to the attendant. The call is presented to the attendant or placed in the attendant queue.

Within a network environment, the Call Waiting Recall and Camp-on Waiting Recall enhancements have to be configured at a node. Both the Call Waiting Recall and Camp-on Waiting Recall enhancements operate in the same way as in the standalone case. The location of the calling and called party and the attendant have no affect on the call processing.

Network Attendant Service (NAS) is not required, but it may be applied at a node. In this case, NAS takes precedence over the Call Waiting Recall and Camp-on Waiting Recall enhancements, in that the target set is disconnected from the call due to time-out and not to the attendant pressing the Loop key or Recall key.

Operating parameters

The same as for Slow Answer Recall.

Feature interactions

Attendant Recall with Splitting
Multi-Party Operations
Secrecy Enhancement

The Call Waiting Recall and Camp-on Waiting Recall enhancements take precedence over Attendant Recall Splitting (ATS), Secrecy (SYA), Enhanced Secrecy (EHS), and Multiple Party Operations.

Call Waiting Recall
Camp-on Waiting Recall

The Call Waiting Recall and Camp-on Waiting Recall enhancements are compatible with Station Camp-on (STCA).

A forced Camp-on override recall occurs to the attendant. If the Call Waiting Recall and Camp-on Waiting Recall enhancements are equipped, the destination is automatically disconnected when the attendant answers. If the Call Waiting Recall and Camp-on Waiting Recall enhancements are not equipped, and the attendant answers the recall at the same time that the destination answers, a conference is established between the attendant, source, and destination.

Intercept Computer Dial from Directory

If the attendant extends an SRC party to a DEST party on the local node, but slow answer recall occurs since the DEST does not answer, it is possible to dial a new DN from the ICP (the DEST is disconnected when the attendant answers).

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 15 – This enhancement is provisioned on a customer basis.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
...		
- OPT	(SLD) SLA	Slow Answer Recall Enhancement (denied) allowed.

Feature operation

Slow Answer Recall Enhancement

When a Slow Answer Recall occurs the call is placed in the attendant queue and appears on the console. The target set will continue to ring while the recall is queued and presented on the console but unanswered. When the attendant answers the recall, by pressing the appropriate **Loop** key or the **Recall ICI** key, the target set will be disconnected as soon as the Attendant Console answers the Slow Answer Recall.

Call Waiting Recall

Until the attendant answers the call, it remains waiting on the target set, and can still be answered. If the attendant answers the recall by pressing the appropriate **Loop** key or the **Recall** key, the target set is disconnected and can no longer answer the call – the target set will have to be redialed to extend the call.

Camp-on Recall

Until the attendant answers the call, the call remains camped-on to the target set, and can still be answered. If the attendant answers the recall by pressing the appropriate Loop key or the Recall key, the target set is disconnected and can no longer answer the call; the target set will have to be redialed to extend the call.

Introduced in X11 Release:	15.59F
Networking:	No

Slow Answer Recall for Transferred External Trunks

This feature allows an external call to be transferred to a ringing set anywhere within an Integrated Services Digital Network (ISDN) network. The transferred call may be incoming or outgoing, supervised or unsupervised. If the call is not answered within a customer-defined period of time, it is routed to the local attendant as a slow answer recall.

Within a standalone environment, this capability is provided by the Multi-Party Operation feature.

An external call is a call originated by the Public Switched Telephone Network (PSTN). This includes calls originating on a Central Office (CO), Foreign Exchange (FEX), Direct Inward Dialing (DID), or Wide Area Telephone Service (WATS) trunk on a local or remote node, and calls from the PSTN to an ISDN node using Network Attendant Service (NAS) signaling protocol over an ISDN TIE trunk.

Operating parameters

This feature applies only to Meridian 1 systems using Meridian Customer Defined Networking (MCDN) signaling over ISDN Signaling Link (ISL)/ISDN TIE links.

All network nodes must be configured with Network Attendant Service (NAS).

Feature interactions

AC15 Recall: Transfer from Norstar

In both standalone and Network Attendant Service (NAS) environments, when a call is transferred to a ringing set on the Meridian 1 by an AC15 trunk, the RTIM recall timer is not started.

Attendant Recall

Slow Answer Recall Modification (SLAM) has an interaction after the attendant answers the recall. If SLAM is configured, the target set is disconnected after the attendant answers the recall. If SLAM is not configured, the target set rings until the attendant releases it.

Call Forward No Answer

If the ringing station to which the call has been transferred has Call Forward No Answer active, the call will be transferred to the call forward DN after the specified number of ring cycles.

ICP Network Screen Activation, Flexible DN, Meridian Mail Interactions

When an Intercept Computer (ICP) position set transfers an external call across an ISDN network, the slow answer recall timer is set at the transferring node to prevent the terminating set to be rung indefinitely. When the slow answer recall timer times out, the transferred call is recalled to the attendant at the transferring node.

Multi-Party Operations

The Multiple Party Operation recall can only be applied in a standalone environment, and therefore does not interact with this feature.

Network Attendant Service Anti-tromboning

NAS Anti-tromboning is supported by this feature.

Feature packaging

— International Supplementary Features (SUPP) package 131

— Integrated Services Digital Network (ISDN) package 145, **or**

— ISDN Signaling Link (ISL) package 147

— Network Attendant Service (NAS) package 159.

Feature implementation

LD 15 – Configure Slow Answer Recall for Transferred Trunks.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	CDB TIM	Customer Data Block. Release 21 gate opener.
...		
- RTIM	xxx yyy zzz	Recall timers for Slow Answer, Camp-on and Call Waiting, where: xxx = 0-(30)-378 for Slow Answer yyy = 0-(30)-510 for Camp-on, and zzz = 0-(30)-510 for Call Waiting. These timers indicate in seconds the elapsed time before attendant recall. Slow Answer must be a multiple of six seconds. To change one timer, all three fields must be input.

Feature operation

No specific operating procedures are required to use this feature.

Source Included when Attendant Dials

This feature provides a new option in LD 15, which allows the customer to define whether or not the source is to be included in a call while the attendant is dialing the destination (SIAA = allow, SIAD = deny). If the destination answers while the attendant is still included in the call, intrusion tone is provided to all parties to indicate that a conference has been established. The intrusion tone is defined in LD 56, and is a prerequisite for the Source Included when Attendant Dials feature.

If SIAA has been defined, the source will be included in all situations, regardless of the state of the destination, except when the attendant is performing Break-In to a busy station.

The following table outlines the operation, if SIAA has been defined, according to the state of the destination party:

	Source	
Destination	**Included**	**Excluded**
Idle extension	x	
First Degree Busy	x	
Second Degree Busy	x	
Camp-on	x	
Intercept forwarded	x	
Line lock-out	x	
Vacant	x	

Destination	Source	
	Included	Excluded
Busy, Attendant Break-in		x
Meridian Mail	x	
Recorded Announcement	x	
Music	x	

Operating parameters

There are no feature requirements.

Feature interactions

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature will follow the current Source Included when Attendant Dialing handling occurs.

Attendant Break-In

The operation of the Break-In feature is not affected, except that the source receives busy tone before the attendant presses the Break-In (BKI) key.

Attendant Supervisory Console

While the attendant dials the destination, the source receives intrusion tone.

Automatic Call Distribution

The source is included in a conference involving the attendant, the source, and Automatic Call Distribution (ACD). When the call is answered by the ACD agent, intrusion tone is provided to all parties in the conference.

Camp-On
Semi-automatic Camp-On

The source remains included while the attendant dials the destination.

Intercept treatment

If the attendant dials a destination which is intercepted, the source remains included in the call.

Meridian Mail

The source is included in a conference involving the attendant, the source, and Meridian Mail answering. When the call is answered by Meridian Mail, the attendant and source receive intrusion tone.

Recorded Announcement
Music

The source is included in a conference involving the attendant, the source, and Recorded Announcement or music treatment. Intrusion tone is not provided in this case.

Secrecy Enhancement

Source Included when Attendant Dials takes precedence over Secrecy and Enhanced Secrecy.

Feature packaging

— International Supplementary Features (SUPP) package 131

— Flexible Tone and Cadences (FTC) package 125

— Trunk Barring (TBAR) package 132

Feature implementation

LD 15 – Allow or deny Source Included when Attendant Dials for a customer.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add, or change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
...		
- OPT	(SIAD) SIAA	(Deny) or allow Source Included when Attendant Dials.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	15.59F
Networking:	No

Special Dial Tones after Dialed Numbers

This feature allows special dial tones to be provided after certain telephone numbers are dialed. Both the telephone numbers and associated dial tones are customer-defined in LD 56. The system can handle a list of up to 20 telephone numbers with a maximum length of five digits. A tone can be associated with each number. Several different tones can be provided during a dialing sequence by defining a tone with any combination of digits in the dialed number. For example, for the number 12345, a tone can be provided after the digit 1 is dialed, after the digits 123 are dialed, and after the whole (12345) number is dialed. This is done by defining a tone with the digit 1, a tone with the digits 123, and a tone with the digits 12345.

When a number is dialed, the system performs digit analysis. As soon as the dialing sequence is recognized as part of the customer-defined list, the system provides the associated tone, if one has been defined. The tone is generated after all other treatment of digits is performed. As soon as another digit is dialed, the tone is removed. This digit analysis is done until the dialing sequence is completed.

Tones are provided to the following originating terminals:

— all types of sets (including data terminals) and attendants, and

— TIE trunks, except those with MFC/MFE signaling.

Operating parameters

The system performs digit analysis before any other treatment of digits, except digit insertion for incoming trunk calls.

In a network environment, digit recognition is reported to the distant node, which must be equipped to handle the processing.

Feature interactions

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

The Special Dial Tones after Dialed Numbers feature is supported in a DPNSS1 UDP network.

Digital Trunk Interface (DTI) – Commonwealth of Independent States (CIS)

Special Dial Tones can be used to provide dial tone after the Meridian 1 user has dialed the digit "9" (Local Exchange access code).

EuroISDN Master Mode

This feature is not supported for incoming calls on the ETSI network side, but it is supported for outgoing calls.

Special dial tone after access codes

Special dial tone after access codes takes precedence over the special dial tones after dialed number treatment. To define special dial tones after access codes, NO has to be entered in response to prompt DLTN in LD 86 (to inhibit dial tone to access codes). The access code digits and associated tones would then have to be defined in response to the DTAD prompt in LD 56.

Feature packaging

Flexible Numbering Plan (PNP) package 160; and to define SRC1-SRC8 special tones, the Flexible Tones and Cadences (FTC) package 125.

Feature implementation

LD 86 – To define special dial tones after access codes, enter NO to the DLTN prompt (to inhibit dial tone to access codes).

Prompt	Response	Description
...		
DLTN	(YES) NO	NARS/BARS Dial Tone after dialing AC1 or AC2 access codes.

LD 56 – Configure Special Dial Tones after Dialed Numbers.

Prompt	Response	Description
...		
TYPE	DTAD	Special Dial Tone after Dialed Number data block.
...		
DDGT	xxxxx	Dialed digits (1-5 digits).
	X	To remove.
	<CR>	To end.
- TONE	(DIAL) SPDT, SRC1-SRC8	Enter a value of DIAL (for dial tone), SPDT (for special dial tone), or SRC1-SRC8 (for other types of tone). For the latter entry, the Flexible Tones and Cadences (FTC) package 125 must be equipped.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	14.47E
Networking:	No

Special Signaling Protocols

This feature allows the existing Swedish analog (500/2500 type) telephones to be connected through analog TIE trunks to the Meridian 1. These TIE trunks use Swedish signaling protocols. The TIE trunks can be divided into the following types:

— automatic

— semi-automatic

— tone, or

— Automatic Telephony (ATL) (when the Swedish ATL trunk support feature is equipped).

Operating parameters

The Swedish TIE trunk types do not apply to digital TIE trunks.

The Swedish TIE trunk types cannot be mixed on a route.

The Swedish TIE trunks require trunk cards of type TPC71 or TPC237. The trunk cards must be placed on specific Televerket (TVT) loops.

A semi-automatic or tone TIE trunk should not be connected to another Meridian 1 trunk. An incoming Public Exchange/Central Office trunk can be connected to an outgoing automatic TIE trunk.

Feature interactions

There are no interactions with other features.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Create or modify data for trunk routes:

Prompt	Response	Description
...		
TKTP	TIE SEMI TIE AUTO TIE TONE	Semi-automatic TIE trunk data block. Automatic TIE trunk data block. Tone TIE trunk data block.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	15.58F
Networking:	No

Special Trunk Support

This feature allows the interface of the Meridian 1 with the Swedish Automatic Telephony (ATL) military radio-link network.

Operating parameters

ATL trunks must never be used for tandem switching or for networks using Electronic Switched Network (ESN) proprietary signaling.

Echo suppression and loss adjustment cannot be effected through software change.

Modified TPC237 cards must be used for ATL trunks, and must be configured on loops specifically defined for Televerket (TVT) use. An SSO adapter is used between the ATL network trunk and the TPC237 card.

Feature interactions

There are no interactions with other features.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 14 – Respond to the following prompts.

Prompt	Response	Description
...		
TYPE	TIE	TIE Trunk data block.
TN	l s c u c u	Terminal Number. For Option 11C.
...		
CUST	0-99 0-31	Customer number. For Option 11C.
...		
NCOS	(0)	Network Class of Service.
RTMB	0-511 1-510 0-127 1-510	Roue number, Member number. For Option 11C.
...		
MNDN	9	Manual Directory Number.
TGAR	(0)	Trunk Group Access Restriction.
SIGL	EAM	Trunk signaling. E&M two-wire.
...		
STRI	WNK	Wink or Fast Flash.
STRO	WNK	Wink or Fast Flash
SUPN	YES	Answer and disconnect supervision required.

LD 16 – Respond to the following prompts.

Prompt	Response	Description
...		
TYPE	RDB	Route data block.
CUST	0-99 0-31	Customer number. For Option 11C.
...		
ROUT	0-511 0-127	Route number. For Option 11C.
TKTP	TIE ATL	The ATL data block for Sweden.
...		
ICOG	IAO	Incoming and outgoing trunk.
...		
SRCH	RRB	Round Robin Hunting for outgoing trunk (start with the next lower trunk than the one seized).
...		
ACOD	xxxx	Access Code for the trunk route. The ACOD must not conflict with the numbering plan.
...		
CNTL	YES	Change controls or timers.
- TIMR	ODT 8064	End of dial tone for Digitone trunks in milliseconds.
- TIMR	EOD 8064	End of Dial, non-Digitone trunks in milliseconds.
- TIMR	DSI 20096	Disconnect Supervision in milliseconds.

- TIMR	ICF 896	Incoming Flash in milliseconds.
- TIMR	OGF 896	Outgoing Flash in milliseconds.
- TIMR	GTI 1152	Incoming Guard in milliseconds.
- TIMR	GTO 1152	Outgoing Guard in milliseconds.
- TIMR	OBA 120	Outgoing B-Answer. Time in seconds to wait for B-Answer on outgoing ATL trunks for Sweden.
- SST	4	Seizure Supervision Timer, in seconds.
NEDC	ETH	Either end control.
FEDC	ETH	Far End Disconnect Control. Either end.
...		
PANS	YES	Pseudo Answer can be sent on some types of trunks as soon as end of dialing is detected. SUPN in LD 14 should be YES, or PANS = YES has no meaning.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	All
Networking:	No

Speed Call

Speed Call allows you to place calls by dialing a one-, two-, or three-digit code. You can use Speed Call for both internal and external calls. To use Speed Call, Meridian 1 proprietary telephones, and Attendant Consoles can have a Speed Call key/lamp pair.

Analog (500/2500 type) telephones can activate Speed Call by using Special Prefix (SPRE) or Flexible Feature Codes (FFC).

Analog (500/2500 type) telephones, Meridian 1 proprietary telephones, and Attendant Consoles can be designated as a Speed Call Controller (SCC) or a Speed Call User (SCU). SCCs can program the numbers to be stored (Speed Call codes) and can use the Speed Call list. SPUs cannot program Speed Call codes; they can only use the Speed Call lists.

Each stored number is assigned a Speed Call code from the Speed Call list. Each list can contain up to 1000 telephone numbers (entries). The maximum number of digits of the telephone number that can be stored in each entry is specified by the customer. Speed Call entries can be 4, 8, 12, 16, 20, 24, 28, or 31 digits long.

Operating parameters

You can define up to 255 (0-254) Speed Call lists per system. X11 Release 13 and later software allows up to 8191 (0-8190) Speed Call lists per system, as long as sufficient memory is available. The new maximum includes all combined Speed Call, System Speed Call (SSC), and Hot Line lists.

You can have as many Speed Call lists as you have available key/lamp pairs on any Meridian 1 proprietary telephone, or Attendant Console. Any number of users can be assigned to a list. Analog (500/2500 type) telephones can access only one Speed Call list. More than one Speed Call Controller can be assigned to each list, but this is not recommended.

A maximum of 31 digits for the telephone number is allowed per Speed Call list entry. An asterisk (*), which indicates a pause, and an octothorpe (#), which indicates end-of-dialing, can be programmed as part of the entry.

Speed Call list entries can be defined in LD 18 or by Speed Call Controllers. Speed Call Controllers must know the digit length (one, two, or three) required for the Speed Call codes in each list.

Feature interactions

AC15 Recall: Transfer from Meridian 1

Speed Call and Network Speed Call are supported with the AC15 Recall: Transfer from Meridian 1 on the first transfer, provided that the digits are outpulsed on the trunk after the End-to-End Signaling Delay timer expires. If the far end is not ready, the call will fail because no dial tone is detected by the Meridian 1.

Additional transfers are supported if the digits are outpulsed without any treatment. For example, the route access code will be outpulsed to the far end. No dial tone detector is assigned and no timer is started so the digits are outpulsed immediately without checking the state at the far end.

Autodial Tandem Transfer

The Speed Call key cannot be used after a Centrex Switchhook Flash or during an established call to send digits out to the far site. The Speed Call key can be used only during the dialing stage.

Automatic Redial

The Automatic Redial (ARDL) feature can be activated on a call using Speed Call (SCL).

Call Forward/Hunt Override Via Flexible Feature Code

The Call Forward/Hunt Override FFC cannot be stored in a speed call list

Call Park

Speed Call can be programmed to parked calls or access parked calls.

Call Party Name Display

No name information displays during the programming of Speed Call numbers.

Calling Party Privacy

An outgoing trunk call initiated by dialing the Speed Call code will carry the Privacy Indicator if the Calling Party Privacy (CPP) code followed by the normal dialing sequence is stored in the Speed Call Entry represented by the Speed Call code. The CPP code will be counted against the maximum number of digits (currently 31) allowed per Speed Call list entry.

A user can also store the CPP code in the Speed Call Entry (or Speed Call key). An outgoing CPP call can then be initiated by dialing the Speed Call code (or pressing the Speed Call key), followed by manually dialing the digits.

However, existing Speed Call limitations do not allow a user to dial *67 (or anything else) before accessing a Speed Call list entry.

Charge Account and Calling Party Number

Charge account numbers, including the Charge Account access Special Prefix (SPRE) code, can be stored as Speed Call or Autodial numbers. All current limitations of these features apply, such as a maximum of 23 digits per entry, including the access code. An Autodial number or dialed digits can follow, but not precede, a Speed Call number. The digits generated by an Autodial key during feature operation are accepted as Charge Account digits.

Charge Account, Forced

Forced Charge Account numbers (including the Special Prefix [SPRE] code and the Charge Account access code) can be entered in Speed Call lists or stored as Autodial numbers. The digits can also be stored, provided that the account number, regardless of its length, is followed directly by an octothorpe (#).

China – Flexible Feature Codes - Outgoing Call Barring

Digits dialed using Speed Call are checked against the active OCB level. This includes calls made using the Dial Access to Speed Call feature (that is, using Pilot DNs).

China Number 1 Signaling Enhancements

Delay Digit Outpulsing will be denied when dialing is done by way of Speed Call.

Direct Private Network Access

If a Speed Call entry is programmed with a valid Authcode for Authcode Last followed by an octothorpe "#", the existing Authcode Last operation will reject the Authcode as an invalid Authcode. If Authcode Last Retry is defined, the caller will be reprompted for the Authcode.

Last Number Redial

A number dialed using Speed Call will become the Last Number Redial number on all telephones, except the M2317 and M3000.

Pretranslation

With X11 Release 18 and later, a Speed Call List number should be programmed to allow for Pretranslation. For example, if 9 pretranslates to 99 and you want to reach 99 nxx xxxx, you need to program the number in the Speed Call List as 9 nxx xxxx. When the Speed Call List is used, 9 nxx xxxx is pretranslated at call processing time to become 99 nxx xxxx.

If Pretranslation is enabled for a customer, then when a Speed Call List is assigned to a Pretranslation group within the customer, it cannot be accessed by a Meridian 1 proprietary set from within that customer group.

Scheduled Access Restrictions

The System Speed Call features ignore the Class of Service and TGAR access restrictions in a Scheduled Access Restriction schedule, using the Class of Service and NCOS defined in the speed call list.

Speed Call Delimiter

An octothorpe (#) is required as a delimiter following an authorization code if an Electronic Switched Network (ESN) and dialed number are stored as part of the speed call or autodial key. If an octothorpe (#) is not entered then the user receives a fast busy tone. If the MSCD = YES, then the end of dial delimiter must be programmed to something other than an octothorpe (#) in LD 15.

Speed Call Directory Number Access

Speed Call DN Access is an enhancement of the Speed Call List (SCL) and System Speed Call (SSC) List features. Refer to SCL and SSC feature descriptions for interactions with other features.

Station Specific Authorization Code

Station Specific Authorization Code (SSAU) feature treats stored autodial numbers as if they were entered at the telephone.

Three Wire Analog Trunk – Commonwealth of Independent States (CIS)

Speed Call on an E3W trunk will fail for toll calls. E3W trunks do not wait for the ANI request from the Public Exchange, that is expected to appear after the toll access code is dialed. The Public Exchange will not accept the call due to the failure to receive ANI information.

User Selectable Call Redirection

Speed Call is not supported by User Selectable Call Redirection.

Feature packaging

Speed Call is part of Optional Features (OPTF) package 1, and has no feature package dependencies.

Feature implementation

LD 17 – Set maximum number of Speed Call lists.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PARM	Configuration Record. Release 19 gate opener.
...		
- MSCL	0-8191	Maximum number of Speed Call lists.

LD 18 – Compute Speed Call list memory size and disk records (X11 Release 17). Use this prompt sequence to determine if there is enough memory and disk records for new Speed Call lists. Compare the output with the MEM AVAIL and DISK AVAIL values output before the REQ prompt.

Prompt	Response	Description
REQ	COMP	Compute disk and memory.
TYPE	SCL	Speed Call lists.
NOLS	1-8191	Number of lists to be added.
DNSZ	4-(16)-31	Maximum length of DN allowed for Speed Call list.
SIZE	1-1000	Maximum number of entries in Speed Call list.

LD 18 – Add or change a Speed Call list.

Prompt	Response	Description
REQ	NEW CHG OUT	Add, change, or remove a Speed Call list.
TYPE	SCL	Speed Call data block.
LNSO	0-8190	Speed Call list number.
DNSZ	4-(16)-31	Maximum number of digits in a list entry (i.e., 4, 8, 12, 16, 20, 24, 28, or 31).
SIZE	1-1000	Maximum number of entries in the Speed Call list.
WRT	(YES) NO	Data is correct and list may be updated.
STOR	xxx	xxx = list entry number (0-9, 00-99, or 000-999).
	yy...yy	yy = digits to be stored against the entry (must be equal to or less than DNSZ).
WRT	(YES) NO	Data is correct and list can be updated.

Note: The prompt WRT follows prompts SIZE and STOR, asking you to confirm the correctness of the data just entered. If data is correct, enter YES or <CR>. A response of NO after the SIZE prompt causes all data entered to be ignored. A response of NO after the STOR prompt generates a warning message (SCH3213) indicating the data was not stored and must be reentered.

A response of **** aborts the program. Only the last STOR value is lost. All previous values to which WRT was YES are saved.

In X11 Release 17 and later, the following information is output with the WRT prompt, following SIZE:

ADDS: MEM: xxxxx DISK: yy.y

where xxxxx is the amount of protected memory and yy.y is the number of disk records required for the new Speed Call list. Check the MEM AVAIL and DISK REC AVAIL values output before the REQ prompt.

LD 10 – Add or change Speed Call for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
FTR	SCU yyyy	Speed Call User, list number (0-8190).
	SCC yyyy	Speed Call Controller, list number (0-8190).

LD 11 – Assign a Speed Call list to Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx SCU yyyy	System Speed Call User key.
	xx SCC yyyy	Speed Call Controller key, where: xx = key number, and yyyy = Speed Call list number (0-8190). M3000 must use key 21. M2317 must use key 0-10 or key 21.

LD 12 – Assign a Speed Call list to an Attendant Console.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT 1250 2250	Console type.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx SCC yyyy	Speed Call Controller, where: xx = key number, and yyyy = list number (0-8190).

Feature operation

To store Speed Call entries from a Meridian 1 proprietary telephone, or Attendant Console (Controller):

1. Without lifting the handset, press **Speed Call.** The indicator flashes.
2. Dial the Speed Call code (0-999), followed by the phone number it represents.
3. Press **Speed Call.** If the entry is accepted, the indicator goes off. If the entry is not accepted, the indicator continues flashing.

To make a Speed Call from a Meridian 1 proprietary telephone, or Attendant Console (User):

1. Lift the handset and press **Speed Call** (telephone).
   - Select an idle loop key and press **Speed Call** (Attendant Console).
2. Dial the Speed Call code. The telephone number represented by the Speed Call code is dialed automatically.

To store Speed Call entries from an analog (500/2500 type) telephone (Controller):

1 Lift the handset and press octothorpe (#) +2 (2500 telephone) or SPRE+75 (analog (500/2500 type) telephone).

2 Dial the Speed Call code (0-999), followed by the phone number it represents. If the entry is accepted, you hear silence. If the entry is not accepted, you hear a fast busy tone.

3 Hang up.

Repeat steps 1 through 3 for each entry to be stored.

To make a Speed Call from an analog (500/2500 type) telephone (User):

1 Lift the handset and dial #3 (2500 telephone), or SPRE 76 (analog (500/2500 type) telephone).

2 Dial the Speed Call code (0-999). The telephone number represented by the Speed Call code is dialed automatically.

Note: In addition to SPRE codes your system may be equipped with Flexible Feature Codes (FFCs).

Introduced in X11 Release:	13
Networking:	No

Speed Call/Autodial with Authorization Codes

This feature is an enhancement to the existing Speed Call and Autodial features. It allows a Speed Call entry to contain an Authorization Code with an associated trunk route or Electronic Switched Network (ESN) access code and dialed number. The digits stored are recorded in Call Detail Recording (CDR), if equipped, for billing purposes.

The Speed Call entry can be one of the following:

— SPRE + 6 + Authorization Code

— SPRE + 6 + Authorization Code + #, or

— SPRE + 6 + Authorization Code + # + ESN access code and dialed number.

Operating parameters

Authorization Code Conditionally Last is not supported.

An octothorpe (#) is required as a delimiter after the Authorization Code if an ESN access code and dialed number are stored as part of the Speed Call or Autodial key. If the octothorpe is not entered, the user receives a fast busy tone. The octothorpe is not stored in the CDR record.

If the system initializes before the Authorization Code is recorded by CDR, the record may be lost.

An SL-1 digital display set can display up to 16 digits. Additional digits cause the digits to scroll off the display.

The M3000 set can display up to 29 digits. Additional digits cause the digits to scroll off the display. Only one softkey, key 21, can be programmed for Speed Call.

An M2317 set can display up to 31 digits.

For Meridian 1 proprietary telephones, up to 31 digits per Speed Call entry are allowed.

On digit display sets, Authorization Codes cannot be blocked from being displayed.

There is no validation of the Authorization Code until the Speed Call key is activated.

Feature interactions

There are no interactions with other features.

Feature packaging

The following packages are required to implement this feature:

- Basic Authorization Code (BAUT) package 25, or Network Authorization Code (NAUT) package 63.
- Optional features (OPTF) package 1, System Speed Call (SSC) package 34, or Network Speed Call (NSC) package 39.

Feature implementation

An Authorization Code can now be entered as part of a Speed Call list.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	22.0X
Networking:	No

Speed Call Delimiter

The Speed Call Delimiter feature meets the Chinese Ministry of Posts and Telecommunications requirements for the operation of Speed Call and System Speed Call. This feature operates similar to the Speed Call and System Speed Call with the exception of delimiters and confirmation tones.

The Speed Call Delimiter feature requires a Speed Call controller to enter an asterisk (*) between abbreviated numbers and telephone numbers when configuring speed call lists. An additional octothorpe (#) delimiter is required for Analog (2500-type) sets to indicate the end of dialing. If an octothorpe (#) is not entered, then the telephone number is not stored and the entry is not valid.

The octothorpe (#) delimiter has the flexibility of being programmed as mandatory or optional. The delimiter can be modified to something other than an octothorpe (#).

Operating parameters

An asterisk (*) delimiter is used when programming speed call lists only. An asterisk (*) can also be used as a three second delay.

No changes occur when a user wants to display a number stored against a list entry number. To display a stored entry the user presses the Display key and the Speed Call key and dials the list number. The list number cannot be abbreviated.

This feature does not apply to Analog 500-type telephones.

The use of confirmation tone or announcement implies the use of an (#) as end of dial speed call delimiter. This means that an (#) cannot be stored as part of the digit string.

An octothorpe (#) is required as a delimiter following an authorization code if an Electronic Switched Network access code and dialed number are part of the Speed Call or Autodial Key. If the (#) is not entered, then the user receives a fast busy tone. Therefore if MSCD = YES, then the end of dial delimiter must be programmed to something other than an octothorpe at the FFCS prompt in LD 15.

Feature interactions

Autodial
Speed Call

An octothorpe (#) is required as a delimiter following an authorization code if an Electronic Switched Network (ESN) and dialed number are stored as part of the speed call or autodial key. If an octothorpe (#) is not entered then the user receives a fast busy tone. If the MSCD = YES, then the end of dial delimiter must be programmed to something other than an octothorpe (#) in LD 15.

Group Call List

Speed Call Delimiter does not interact with Group Call List.

Outpulsing Asterisk (*) and Octothorpe (#)

If the Outpulsing Asterisk (*) and Octothorpe (#) (OPAO) package 104 is equipped and the configuration tone is programmed, then the value stored in the STRG prompt (LD 15) is entered rather than an octothorpe (#) to indicate the end of dial string. Following this, the numbers are stored.

Feature packaging

China Speed Call Delimiter requires Speed Call (OBTF) package 1 and System Speed Call (SSC) package 34.

Flexible Feature Codes (FFC) package 139 is required for Analog 2500-type telephones, if a set accesses speed call list or system speed call list or attendant console. This package is optional for Meridian 1 proprietary sets or Attendant Consoles because these sets can access Speed Call List/System Speed Call List by using a key.

Feature implementation

To enable Speed Call and System Speed Call, the maximum number of speed call lists must be determined in LD 17. The speed call list memory size must also be configured in LD 18. For more information on these overlays and the assignment of these features to Meridian 1 proprietary, analog (500/2500-type) telephones and attendant consoles please refer to the sections entitled Speed Call and System Speed Call in this publication.

LD 15 - Enable Speed Call Delimiter in Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	FTR	Flexible Feature Code gate opener.
...		
CUST	xx	Customer number.
...		
- LEND	YES	List Entry Number Delimiter. If LEND=YES, then an asterisk (*) delimiter between the list entry number and telephone number must be entered. If LEND=NO, then existing Speed Call operation continues.
- MSCD	YES	Mandatory Speed Call Delimiter. Default = Octothorpe (#). An octothorpe (#) is required after entering telephone number to indicate the end of dial. If MSCD=NO, then the end of dial Speed Call Delimiter octothorpe (#) is optional.

Note: The China market requires an octothorpe (#) delimiter at the end of dialing. Other markets have the option of selecting a mandatory or optional delimiter by entering "YES" or "NO" at the MSCD prompt. The end of dial delimiter can be an octothorpe (default value) or it can be changed to another delimiter by modifying values at the Flexible Feature Code end-of-dialing indicator (FFCS). String to indicate end-of-dialing (STRG) and string length of end-of-dial indictor (STRL) prompts in LD 15.

Feature operation

Speed Call Delimiter Operation

Analog 2500-type telephone

1 To Program Speed Call List - Go off-hook, dial and receive dial tone. Dial System Speed Call Controller (SCC) FFC code, the list entry number and telephone number (for example, *51*1*5556667777#). Get response. If accepted, then confirmation tone or announcement is configured and the end of dial speed call delimiter is entered. Response is a tone or speech signal. Otherwise, silence is given. Go on-hook.

2 To Use - Go off-hook and receive dial tone. Dial Speed Call User (SCU) code and the list entry number.

3 To Delete List Entry Number in Speed Call List - Go off-hook and receive dial tone. Dial Speed Call Erase (SCE) FFC followed by list entry number (0 - 999) and (#) delimiter. Delete the specific list entry number.

Meridian 1 Proprietary telephones and Attendant Console

1 To Program Speed Call List – Press Speed Call Controller Key and the indicator flashes. Dial list entry number (0 - 9999) followed by an asterisk (e.g. 1*5556667777). Press Speed Call Controller Key again. If entry is accepted, the indicator goes off. If the entry is not accepted, then the indicator remains flashing. An asterisk is only used to indicate the end of dial of list entry number and is not stored as a digit string.

2 To Use on Meridian 1 Proprietary Telephones – Lift the handset or press DN Key. Press System Speed Call Controller (SCC) or Speed Call User (SCU) Key. Dial the list entry number.

3 Use Attendant Console – Press an idle loop key and then press Speed Call Controller (SCC) Key. Dial the list entry number.

System Speed Call Delimiter Operation

Meridian 1 Proprietary telephones

1 To Program System Speed Call List – Press assigned System Speed Call Controller Key and indicator flashes. Dial list entry number (0 - 999) followed by an asterisk (*) and then the telephone number (for example, 1*0115556667777). Then press SSC/SSU Key again. If accepted, the indictor goes off. If not accepted, the indicator remains flashing.

2 To Use - Lift handset or press DN key. Press SSC/SSU Key. Dial the list entry number, or lift handset or press DN key. Dial SSU FFC code. Dial list entry number.

Attendant Console

1 To Program System Speed Call List - Press SSC Key and indicator flashes. Dial list entry number (0 - 999), followed by an asterisk (*) and the telephone number. Press SSC Key again. If entry is accepted, indicator goes off. If the entry is not accepted, indicator continues to flash.

2 To Use - Press an idle loop Key. Dial SSU FFC code. Dial list entry number.

Introduced in X11 Release:	15.58F
Networking:	No

Speed Call Directory Number Access

The Speed Call Directory Number (DN) Access feature allows a Pilot DN to be used as an access code to either a Speed Call List (SCL) or a System Speed Call List (SSC).

Speed Call DN Access provides an alternative way to access either a Speed Call List or a System Speed Call List. Instead of dialing the Special Prefix (SPRE), a SCL or SSC access code, and a list entry number, or instead of depressing an idle DN key, a SCL or SSC key, and then dialing a list entry number, a user can alternatively dial a speed call access Pilot DN followed by the list entry number.

Since each speed call access Pilot DN is associated with a SCL or SSC list, users can access as many SCL or SSC lists as they need by dialing the appropriate Pilot DN.

A Pilot DN can be accessed from anywhere in a network, so that any network user can access all speed call lists defined for a network, from anywhere in the network. This allows a centralized Speed Call List to be set up for the entire network.

Operating parameters

The requirements for Speed Call and System Speed Call also apply to this feature.

Feature interactions

Direct Inward Dialing (DID) and TIE trunk access

An additional one to three digits will be accepted from these trunks to complete a Speed Call, provided these additional digits are allowed to be sent by the external system.

Speed Call
System Speed Call

Speed Call DN Access is an enhancement of the SCL and SSC features. Refer to SCL and SSC feature descriptions for interactions with other features.

Feature packaging

Speed Call Directory Number Access requires Group Hunt/DN Access to SCL (PLDN) package 120.

Dependencies:

— International Supplementary Features (SUPP) package 131

— Flexible Feature Codes (FFC) package 139

— System Speed Call (SSC) package 34

— Optional Features (OPTF) number 1

Feature implementation

LD 57 – Define, change, print, or remove data associated with FFC. A new PLDN prompt is introduced for Pilot DNs.The new LSNO prompt is used to associate the Pilot DN with a SCL or SSC list. The USE prompt is displayed only if the Pilot DN entered in response to the PLDN prompt has not already been defined.

Prompt	Response	Description
REQ	CHG NEW	Modify or create data block.
TYPE	FFC	Flexible Feature Codes data block.
CUST	0-99 0-31	Customer to which the data block belongs. For Option 11C.
FFCT	<CR>	Flexible Feature Confirmation Tone.
CODE	PLDN	Code to be modified or created: Pilot DN.
PLDN	xxxx <CR>	Pilot DN: enter Pilot DN to be modified or created; enter carriage return to proceed to next prompt.
USE	 SCLC SCLU	USE: enter USE for Pilot DN. Speed Call List Controller. Speed Call List User.
LSNO	xxxx	List Number: enter Speed Call or System Speed Call list number. Speed Call list must exist in LD 18.

Prompt	Response	Description
REQ	OUT PRT	Remove or print a code or data block.
TYPE	FFC	Flexible Feature Codes data block.
CUST	0-99 0-31	Customer to which the data block belongs. For Option 11C.
CODE	PLDN ALL	Code requested: Pilot DN. All FFC.
PLDN	xxxx <CR>	Pilot DN: enter Pilot DN to be removed enter carriage return to proceed next prompt

Feature operation

To access either a Speed Call List or a System Speed Call List using this feature, dial a speed call access Pilot DN followed by the list entry number.

Pilot DN

Pilot DNs are defined as PLDN Flexible Feature Codes (FFC) via service change LD 57.

Pilot DNs can be used in two ways:

1. If the USE prompt is set to GPHT, the Pilot DN is defined to activate Group Hunting.

2. If the USE prompt is set to SCLC (Speed Call List Controller) or SCLU (Speed Call List User), the Pilot DN is defined to access the Speed Call or System Speed Call lists that are associated with the Pilot DN.

When the response to the USE prompt is SCLC (controller), a station can modify an SCL or SSC list by dialing the speed call access Pilot DN associated with that list, followed by a one- to three-digit list entry number, the number to be entered in the list, and then going on-hook. Overflow tone is returned if the information entered is not valid. Confirmation tone is returned if the Flexible Feature Confirmation Tone (FFCT) option is set and trailing '#' is dialed, as in existing Flexible Feature Codes (FFCs) operations.

When the response to the USE prompt is SCLU (user), to use any entry in a SCL or SSC list, a station user dials the speed call access Pilot DN associated with the list, followed by the one- to three-digit list entry number.

Speed Call on Private Lines

This feature allows Meridian 1 proprietary telephone users equipped with a Private Line (PVR or PVN) key and a Speed Call (SCL) key to first access a Private Line trunk (by pressing the PVR or PVN key) and then make a speed call (by pressing the SCL key).

Operating parameters

When a Private Line call is made, recognizable Route Access Codes are absorbed from the start of every entry in the Speed Call List (e.g., if 7654 is stored as a Speed Call List entry, and 76 is a valid Route Access Code, 76 is absorbed and 54 is outpulsed).

Feature interactions

For more Feature interactions refer to Private Line and Speed Call features.

Automatic Redial

The ARDL feature is activated on a number dialed using the Private Line (PVR/PVN) key and then making a speed call by pressing the Speed Call (SCL) key.

Basic/Network Alternate Route Selection (BARS/NARS)

The BARS and NARS access codes (AC1 and AC2) are not absorbed. If a user has a Speed Call list entry that includes either AC1 or AC2, this entry will not terminate correctly when used on a Private Line. The BARS or NARS access code (AC1 or AC2) will be outpulsed, causing the Public Network to either terminate the call at an unwanted location or reject the call.

Feature packaging

Speed Call on Private Lines is part of base X11 system software and requires Optional Features (OPTF) package 1.

Feature implementation

No change to existing configuration is required to implement this feature.

Feature operation

	ACTION	RESPONSE
1	User presses Private Line Ringing (**PVR**) or Private Line Nonringing (**PVN**) key.	Trunk is accessed and dial tone is returned.
2	User presses Speed Call key and enters list entry number.	The number stored against this entry is outpulsed.

Introduced in X11 Release:	19
Networking:	No

Speed Call, System

System Speed Call extends the capabilities of Speed Call. In addition to abbreviated dialing, System Speed Call allows a user to temporarily override the telephone's Class of Service, Trunk Group Access Restrictions (TGARs), and code restrictions.

Analog (500/2500 type) telephones, Meridian 1 proprietary telephones, and Attendant Consoles can activate System Speed Call by using SPRE or Flexible Feature Codes (FFC).

An analog (500/2500 type) telephone can be designated as a System Speed Call User only (not Controller) and can access one System Speed Call list. Meridian 1 proprietary telephones can be System Speed Call Users (SPRE codes or key access) or Controllers (key access only). Attendant Consoles can be System Speed Call Users (dial access only) and System Speed Call Controllers (key access only).

Operating parameters

Prior to X11 Release 13 up to 255 (0-254) System Speed Call lists and regular Speed Call lists can be defined per system. X11 Release 13 and later software allows up to 8191 (0-8190) Speed Call lists, as long as sufficient memory is available. The new maximum includes all combined Speed Call, System Speed Call and Hot Line lists, 4096 (0-4095) of which can be System Speed Call lists.

System Speed Call lists can have up to 1000 entries and each entry can be up to 31 digits in length.

Restrictions applied to a telephone are ignored only for the origination of a call made through System Speed Call. Restrictions are applied if any call modification is attempted once the call is established.

System Speed Call lists can only be programmed in LD 18 or from telephones or Attendant Consoles equipped with a System Speed Call Controller key.

Prior to X11 Release 19, the technician enters each System or regular Speed Call List individually. X11 Release 19 enhances LD 18 so the technician can add or copy up to 100 System and regular Speed Call Lists at a time.

Feature interactions

Attendant Administration

System Speed Call lists can be assigned using Attendant Administration.

Authorization Code Security Enhancement

If the Basic Authorization Code (BAUT) or Network Authorization Code (NAUT) package is equipped, a Network Class of Service (NCOS) is assigned to the System Speed Call list. The NCOS of the System Speed Call list replaces the NCOS of the Authorization code or Forced Charge Account code if it increases the Facility Restriction Level (FRL) of the code.

Automatic Redial

The Automatic Redial (ARDL) feature can be activated on a call using System Speed Call (SSU/SSC).

Basic/Network Alternate Route Selection (BARS/NARS)

If the BARS or NARS package is equipped, an NCOS is assigned to the System Speed Call list. The NCOS associated with the System Speed Call list replaces the NCOS of the telephone if it increases the Facility Restriction Level (FRL) of the user.

Calling Party Privacy

An outgoing trunk call initiated by dialing the Speed Call code will carry the Privacy Indicator if the Calling Party Privacy (CPP) code followed by the normal dialing sequence is stored in the Speed Call Entry represented by the Speed Call code. The CPP code will be counted against the maximum number of digits (currently 31) allowed per Speed Call list entry.

A user can also store the CPP code in the Speed Call Entry (or Speed Call key). An outgoing CPP call can then be initiated by dialing the Speed Call code (or pressing the Speed Call key), followed by manually dialing the digits.

However, existing Speed Call limitations do not allow a user to dial *67 (or anything else) before accessing a Speed Call list entry.

Capacity Expansion

Any number from 0 to 4095 can be assigned to a System Speed Call list.

China – Flexible Feature Codes - Outgoing Call Barring

Digits dialed using System Speed Call are checked against the active OCB level.

Flexible Feature Code

With Flexible Feature Code (FFC), a confirmation tone is provided for Speed Call store after the end-of-dial (EOD) string is entered.

Hot Line

When the System Speed Call package is equipped, Hot Line lists have the characteristics and limitations of SSC lists. If the package is not equipped, Hot Line lists function like standard Speed Call lists.

Last Number Redial

A number dialed using a System Speed Call key becomes the Last Number Redial number on all telephones, except the M2317 and M3000. A number dialed using SPRE-activated System Speed Call becomes the Last Number Redial number on all telephones. The original Class of Service and NCOS restrictions of the telephone apply when using Last Number Redial.

Off-Hook Alarm Security

Off-Hook Alarm Security (OHAS) treatment can apply to these features if the ASTM expires. The Alarm Security Timer may expire for the following reasons:

— A dial tone or interdigit timeout occurs while dialing the speed call access code.

— The Speed Call being accessed has an asterisk (*) causing a three-second delay. If the ASTM is three seconds or less, the OHAS intercept treatment may occur.

Pretranslation

With X11 Release 18 and later, a Speed Call List number should be programmed to allow for Pretranslation. For example, if 9 pretranslates to 99 and you want to reach 99 nxx xxxx, you need to program the number in the Speed Call List as 9 nxx xxxx. When the Speed Call List is used, 9 nxx xxxx is pretranslated at call processing time to become 99 nxx xxxx.

Feature packaging

System Speed Call (SSC) package 34 has no feature package dependencies.

Feature implementation

LD 17 – Set maximum number of Speed Call lists.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PARM	Configuration Record. Release 19 gate opener.
...		
- MSCL	0-8190	Maximum number of Speed Call lists.

LD 18 – Compute Speed Call list memory size and disk records (X11 Release 17). Use this prompt sequence to determine if there is enough memory and disk space for new Speed Call lists. Compare the output with the "MEM AVAIL" and "DISK AVAIL" values output before the REQ prompt.

Prompt	Response	Description
REQ	COMP	Compute disk and memory.
TYPE	SCL	Speed Call lists.
NOLS	1-8190	Number of lists to be added.
DNSZ	4-31	Maximum length of DN allowed for Speed Call list.
SIZE	1-1000	Maximum number of entries in Speed Call list.

LD 18 – Add or change a System Speed Call list.

REQ	NEW CHG OUT NEW xx, CPY xx	Add, change, or remove a single speed call list; Add or copy xx lists.
TYPE	SSC SCL	System Speed Call. Speed Call List.
LSNO	0-8190 xxxx yyyy	Number of list to add, where: xxxx = number of list to be copied, and yyyy = number of list to receive copy.
NCOS	0-99	NCOS to be assigned to calls accessing the list.
DNSZ	4-(16)-31	Maximum number of digits in a list entry (i.e., 4, 8, 12, 16, 20, 24, 28, or 31).
SIZE	1-1000	Maximum number of entries in the Speed Call list.
WRT	(YES) NO	Data is correct and list may be updated.
STOR	xxx yy...yy	xxx = list entry number (0-9, 0-99, or 0-999). yy = digits to be stored against the entry (must be equal to or less than DNSZ).
WRT	(YES) NO	Data is correct and list may be updated.

Note: The prompt WRT follows prompts SIZE and STOR asking you to confirm the correctness of the data just entered. If data is correct, enter YES or <CR>. A response of NO after the SIZE prompt causes all data entered to be ignored. A response of NO after the STOR prompt generates a warning message (SCH3213) indicating the data was not stored and must be reentered.

A response of "*****" aborts the program. Only the last STOR value is lost. All previous values to which WRT was YES are saved.

In X11 Release 17 and later, the following information is output with the WRT prompt, following SIZE:

ADDS: MEM: xxxxx DISK: yy.y

Where xxxxx is the amount of protected memory and yy.y is the number of disk records required for the new Speed Call list. Check the "MEM AVAIL" and "DISK REC AVAIL" values output before the REQ prompt.

LD 10 – Add or change System Speed Call for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
FTR	SSU yyyy	System Speed Call user, list number (0-4095).

LD 11 – Add or change System Speed Call list for Meridian 1 proprietary telephones.

REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
SSU	yyyy	System Speed Call list number (0-4095) for dial access.

KEY	xx SSU yyyy xx SSC yyyy	System Speed Call user key. System Speed Call Controller key, where: xx = key number, and yyyy = System Speed Call list number (0-4095). ***Note:*** The M2317 and M3000 must use key 21.

LD 12 – Add or change a System Speed Call list for Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT 1250 2250	Console type.
TN	l s c u c u	Terminal Number. For Option 11C.
SSU	yyyy	System Speed Call list number (0-4095) for dial access.
KEY	xx SSC yyyy	System Speed Call Controller key, where: xx = key number, and yyyy = System Speed Call list number (0-4095).

LD 20 – Print Speed Call data.

Respond to the TYPE prompt with SCL to print regular and System Speed Call lists and pretranslation.

Respond to the TYPE prompt with SSC to print the System Speed Call data block.

Feature operation

To store System Speed Call entries from a Meridian 1 proprietary telephone, or Attendant Console (Controller):

1. Without lifting the handset, press **Speed Call**. The indicator flashes.
2. Dial the Speed Call code (0-999), followed by the telephone number it represents.
3. Press **Speed Call**. If the entry is accepted, the indicator goes off. If the entry is not accepted, the indicator remains flashing.

To make a System Speed Call from a Meridian 1 proprietary telephone, or Attendant Console (User):

1. Lift the handset and dial SPRE 73 or press the System Speed Call key (telephone).

   – or –

   Select an idle loop key and dial SPRE 73 (Attendant Console).

2. Dial the Speed Call code.

   If the Speed Call number is accepted, the telephone number represented by the Speed Call code is dialed automatically. No confirmation tone is given unless Flexible Feature Code (FFC) is implemented.

   If the Speed Call number is not accepted, a fast busy signal indicates the number was rejected.

To make a System Speed Call from an analog (500/2500 type) telephone (User):

1 Lift the handset and dial SPRE 73.

2 Dial the Speed Call code (0-999). The telephone number represented by the Speed Call code is dialed automatically.

Note: In addition to SPRE codes your system can be equipped with Flexible Feature Codes.

The routine to add a call list aborts under the following conditions:

— trying to add a call list whose number is already in use, or

— trying to add multiple call lists when there is insufficient memory.

Introduced in X11 Release:	20.1x
Networking:	Yes

Station Activity Records

When a set is configured with Class of Service Call Detail Monitoring Allowed (CDMA) for all incoming and outgoing calls, Station Activity Records are produced. The format of Station Activity Records is identical to other Call Detail Recording (CDR) records, but they have a new type of identifier (D). Existing CDR records are not affected by this new functionality.

Operating parameters

There are no feature requirements.

Feature interactions

Call Redirection

A Station Activity Record is only produced for a set designated as CDMA that is involved in a call with a trunk. A Station Activity Record is not generated for any set which does not answer the call, regardless of whether it has Class of Service CDMA or CDMD. Any other CDR records generated during call redirection are not affected.

Call Transfer

A Station Activity Record is generated when a set with Class of Service CDMA transfers a trunk call. CDR "X" record generation is not affected by this development. The set to which the call is transferred also produces a Station Activity Record if it has Class of Service CDMA and answers the call. When the second "D" record is produced (by the set to which the call is transferred), the digits field of the "D" record shows the digits dialed by the transferring set.

Conference

For a set with Class of Service CDMA involved in a call with a trunk, a Station Activity Record is produced only when that set conferences in the first party. Conferencing of all subsequent parties does not generate a "D" record. An additional "D" record is produced when the last conferee with Class of Service CDMA connected to the trunk goes on hook. This does not affect any other CDR record generation during a conference.

Internal Call Detail Recording

Internal Call Detail Recording records are produced according to the Class of Service ICDA/ICDD of a set. The Station Activity Record enhancement does not affect the ICDR record generation.

Feature packaging

Station Activity Records is package 251 (SCDR).

Dependencies:

— Call Detail Recording (CDR) package 4

— Call Detail Recording on Teletype Terminal (CTY) package 5

Feature Implementation

LD 10 – Set Class of Service CDMA/CDMD for an analog (500/2500 type) telephone.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	500	Analog (500/2500 type) telephone.
...		
CLS	(CDMD) CDMA	CDMA allows Station Activity Records to be generated for the set (when the trunk is involved in the call). CDMD denies record generation.

LD 11 – Set Class of Service CDMA/CDMD for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	xxxx	Meridian 1 proprietary telephone type.
...		
CLS	(CDMD) CDMA	CDMA allows Station Activity Records to be generated for the set (when the trunk is involved in the call). CDMD denies record generation.

LD 27 – Set Class of Service CDMA/CDMD for BRI sets.

Prompt	Response	Description
REQ	NEW CHG PRT	New, change, or print.
TYPE	DSL	Digital Subscriber Loop.
...		
CLS	(CDMD) CDMA	CDMA allows Station Activity Records to be generated for the set (when the trunk is involved in the call). CDMD denies record generation.

LD 17 – Define a CDR link for Call Detail Recording.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Release 19 gate opener.
USER	CTY	TTY has CTY as the user (for CDR records).

LD 15 – CDR must be enabled for the customer.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	CDB CDR	Customer Data Block. Release 21 gate opener.
CUST	0-99 0-31	Customer number. For Option 11C.
- CDR	YES	Call Detail Recording.
- PORT	0-15	The CDR port number for the customer.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	17
Networking:	No

Station Category Indication

The Station Category Indication (SCI) feature allows an attendant to selectively answer internal attendant Directory Number (DN) calls on a priority basis. Stations are assigned a category, with priority indicated by an Incoming Call Indicator (ICI) lamp at each Attendant Console. Using the answering priority defined in LD 15, the attendant gives prompt attention to a call presented at a high-priority ICI lamp by selecting the associated ICI key.

Operating parameters

A maximum of seven station categories (1-7) can be assigned.

Calls from SCI 0 stations appear on the dial 0 ICI.

Calls from fully restricted stations appear on the dial 0 fully restricted ICI.

The Station Category Indication (SCI) feature should not be mixed with any other Incoming Call Indicator (ICI) assignment on the same ICI key/lamp pair.

Feature interactions

Centralized Attendant Service

When Centralized Attendant Service (CAS) is active, calls from a remote station to the attendant DN appear on the remote ICI key/lamp pair at the CAS main, regardless of the station SCI category.

Controlled Class of Service

The Controlled Class of Service (CCOS) feature has priority over SCI. A station's SCI category is suppressed when CCOS is active, and calls to the attendant DN carry the CCOS class defined in the database.

Phantom Terminal Numbers (TNs)

SCI cannot be enabled on a Phantom TN.

Feature packaging

Station Category Indication (SCI) package 80 has no feature package dependencies.

Feature implementation

LD 15 – Add or change a Station Category Indication ICI key/lamp pair for Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB	Customer Data Block.
CUST	0-99	Customer number.
ICI	0-19 CA1-CA7	Assign ICI key/lamp pair for SCI.
ICI	0-19 DL0	Dial 0 (calls from telephones in SCI 0).
ICI	0-19 DFO	Fully restricted (call from fully restricted telephones).

LD 10 – Change SCI for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
SCI	0-7	SCI number.

LD 11 – Change SCI for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
SCI	0-7	SCI number.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	19
Networking:	No

Station Specific Authorization Code

Station Specific Authorization Code (SSAU) is available with X11 Release 19 and later, and enables the system administrator to control the level of authorization code access on a per telephone basis. SSAU applies to analog (500/2500 type) telephones and Meridian 1 proprietary telephones; it does not apply to Basic Rate Interface (BRI) telephones.

Station Specific Authorization Code provides three levels of authorization code access:

1. Authcode Unrestricted (AUTU)
   An AUTU telephone has no authorization code access limitations. Any authorization code is accepted and processed normally.

2. Authcode Restricted (AUTR)
   An AUTR telephone can enter up to six assigned authorization codes. The authorization code entered must match one of the preassigned codes. Any other authorization code will be rejected and the call will not be completed.

3. Authcode Denied (AUTD)
   An AUTD telephone has no access to authorization codes. Any authorization code will be rejected and the call will not be completed.

Operating parameters

The same authorization code can be assigned to more than one AUTR telephone.

There is cross-checking between LDs 10 and 11, which define a station specific authorization code, and LD 88, which ensures that the user has entered a valid authorization code.

LD 88, which is used to delete an existing authorization code, does not check if the authorization code is assigned as a station specific authorization code before the deletion.

The Station Specific Authorization Code feature does not apply when the authorization code is prompted from a tandem node.

Feature interactions

Attendant Administration

Station Specific Authorization Code does not support Attendant Administration.

Authorization Code Security Enhancement

Users cannot freely enter authorization codes from telephones that have AUTR or AUTD Class of Service.

Autodial
Speed Call

The SSAU feature treats stored autodial numbers as if they were entered at the telephone.

Feature packaging

Station Specific Authorization Code (SSAU) is package 229, which requires Basic Authorization Codes (BAUT) package 25.

Feature implementation

LD 88 – Create Authorization Code data block (AUB).

Prompt	Response	Description
REQ	NEW	Create.
TYPE	AUB	Authcode data block.
CUST	0-99 0-31	Customer number. For Option 11C.
SPWD	xxxx	Secure data password.
ALEN	1-14	Number of digits in authcodes.

ACDR	YES NO	Activate CDR for authcodes. There is no default.
RANR	0-511	RAN route number for "Authcode Last" prompt (NAUT).
CLAS	(0)-115	Class code value assigned to authcode (NAUT).
COS	aaa	Class of Service.
TGAR	(0)-31	Trunk Group Access Restrictions.
NCOS	(0)-99	Network Class of Service.
AUTO	YES, NO	Automatically generate authcodes.
- SECR	0-9999	Security password (NAUT).
- NMBR	1-9999	Number of authcodes to be generated.
- CLAS	(0)-115	Class code value assigned to authcode (NAUT).

LD 88 – Create an Authorization Code Table.

Prompt	**Response**	**Description**
REQ	NEW	Create.
TYPE	AUT	Authorization Code Table.
CUST	0-99 0-31	Customer numbers. For Option 11C.
SPWD	xxxx	Secure data password.
CODE	xxxx	Authcode (number of digits must equal ALEN).
CLAS	(0)-115	Class code value assigned to authcode (NAUT).

LD 10/11 – Activate SSAU.

Prompt	Response	Description
REQ	NEW, CHG	Add, or modify.
TYPE	aaaa	Telephone type, where: aaaa = 500, 2500, SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
CLS	(AUTU) AUTR AUTD	Authcode unrestricted. Authcode restricted. Authcode denied.
MAUT	(NO) YES	Modify assigned authcodes for this telephone.
SPWD	xxxx	Correct security password (if one is defined).
AUTH	x nnnn	x is in the range of 1-6; nnnn is the assigned authcode (a valid authorization code defined in LD 88).
	X x	X x deletes an assigned authcode.

Note: Changing an AUTR telephone to AUTU or AUTD clears all assigned authcode information previously defined for that telephone.

LD 20 - Security Password.

Prompt	Response	Description
REQ	PRT	Print.
TYPE	xxx	Type of data block.
TN	l s c u c u	Terminal Number. For Option 11C.
CDEN	SD DD 4D 8D	Card density requested.
CUST	0-99 0-31	Customer numbers. For Option 11C.

SPWD	xxxx	Valid Security data password to display SSAU.
Note: Once SPWD is prompted, a valid security data password as defined in the customer data block is required for displaying Authorization (AUTH) information for sets with Class of Service Authorization Code. Sets with Class of Service Authcode Unrestricted (AUTU) and Authcode Denied (AUTD) do not have AUTH information for display. Entering of a carriage return at the SPWD prompt will result in the AUTH information being skipped during printing.		

In LD 20, Security Password (SPWD) will not be prompted if any of the following conditions exists:

— the Station Specific Authcode Package 220 is not equipped,

— the response to the TN prompt is more than one specific TN,

— the response to the TN prompt is a unique TN, but the customer of this TN does not have a security data password defined,

— the response to the CUST prompt is not a specific customer, or

— the response to the CUST prompt is a specific customer number but the customer does not have a security password defined.

Feature operation

After an authorization code is entered, the Station Specific Authorization Code feature determines if the set is allowed to use the entered code. If the authorization code is not allowed on that set, the existing invalid authorization code treatment occurs. Otherwise, normal authorization code processing occurs.

Station-to-Station Calling

Station-to-Station Calling allows direct dialing between station users in the same customer group without the assistance of the attendant.

Operating parameters

There are no feature requirements.

Feature interactions

Manual Line Service

If a single line telephone has been assigned a Manual Line Class of Service, the telephone automatically rings the attendant when it goes off-hook.

Private Lines

You must go over the public network to reach a Private Line. The software PRDN is not meant to be dialed directly.

Feature packaging

Station-to-Station Calling is included in base X11 system software.

Feature implementation

No change to existing configuration is required to implement this feature.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	3
Networking:	No

Stored Number Redial

Stored Number Redial (SNR) allows telephones and Attendant Consoles to store one previously dialed number of 4 to 31 digits for automatic redialing.

Depending on the type of telephone, the number can be stored before a call is placed, during Ringback, while the number is busy, or during an active call. On Attendant Consoles, the number can be stored only before a call is placed. Stored Number Redial (SNR) is not supported on M2317 telephones, M3000 Touchphones, or analog (500/2500 type) telephones serving as Private Lines.

Operating parameters

When a number is stored, it overwrites any previously stored number.

Storage is limited to one number per analog (500/2500 type) telephone and one number per SNR key. When a call is established through a Tandem TIE Trunk Network (TTTN), the user is required to pause for dial tone. When you store a number using SNR, automatic redialing may fail because required delays are not added. It is possible to include delays in the outpulsing by dialing the asterisk (*) in the original digit string where dial tone is expected. Each asterisk (*) signifies a three-second delay in outpulsing.

The three-second delay is not available from a 500-type telephone.

During the stored Number Redial (SNR) programming mode, if the user attempts to store more digits than the maximum number defined for the telephone or console, SNR programming is canceled and overflow tone is returned. During an active call on a Meridian 1 proprietary telephone, if a user attempts to store more digits than the specified limit, the SNR operation fails, the previously stored number remains unchanged, and a failure indication is not given. The SNR indicator remains off.

For analog (500/2500 type) telephones, in order to store a number dialed to a busy DN, the maximum length of the stored number must be at least five digits (see prompt FTR RDL xx in LD 10).

Feature interactions

Authorization Code Security Enhancement
Charge Account
Forced Charge Account

The Authorization, Charge Account, and Forced Charge Account codes are not stored. To store a code, dial the code prior to using Stored Number Redial to dial the call.

Automatic Redial

The Automatic Redial (ARDL) feature can be activated on a call using Stored Number Redial (RDL) key.

Calling Party Privacy

During Stored Number Redial (SNR) programming, a user can store the Calling Party Privacy (CPP) code followed by the normal dialing sequence in the SNR data space. Outgoing calls originated by the SNR feature will send the Privacy Indicator to the far end. The CPP code will be counted against the maximum number of digits (currently 31) allowed by the SNR feature.

During an active call on a Meridian 1 proprietary telephone, the Stored Number Redial feature will set a CPP flag in the SNR data space if the CPP code was included in the number dialed by the originator. The outgoing redialed calls will send the Privacy Indicator to the far end.

China Number 1 Signaling Enhancements

Delay Digit Outpulsing will be denied when dialing is done by way of Stored Number Redial.

End-to-End Signaling

End-to-End Signaling (EES) activates after a call to a trunk is established by expiration of the end-of-dial timer. Further digits dialed are not stored by the SNR feature once it is in EES mode.

Group Hunt

A Pilot DN will be stored as a Stored Number Redial (SNR) number when it is dialed directly.

Intercept Computer Dial from Directory - Post-dial Operation

An attendant can dial an extension from the Intercept Computer, and then press the Stored Number Redial key to store the called number (following the rules of the Stored Number Redial feature).

Multi-Party Operations

For analog (500/2500 type) telephones, the Last Number Redial/Stored Number Redial feature can be used when normal or special dial tone is received. The last number redialed that can be stored is the first call of a consultation connection, and can be stored only after the connection is completely released.

Feature packaging

Stored Number Redial (SNR) package 64 has no feature package dependencies.

Feature implementation

LD 10 – Add or change SNR for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	(XFD) XFA	Call Transfer (denied) allowed.
FTR	RDL xx	Activate SNR, where: xx = the maximum number of digits that can be stored (i.e., 4, 8, 12, (16), 20, 24, 28, 31).

LD 11 – Add or change SNR for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, or 2616.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx RDL yy	Add an SNR key, where xx = key number, and yy = the maximum number of digits that can be stored (i.e., 4, 8, 12, (16), 20, 24, 28, 31).

LD 12 – Add or change SNR for Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT 1250 2250	Console type.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx RDL	Add an SNR key.

Feature operation

Attendant Consoles, Meridian 1 proprietary telephones

To store a number prior to dialing (for Attendant Consoles, and Meridian 1 proprietary telephones):

1 Without lifting the handset, press **Stored No.**

2 Dial the number.

3 Press **Stored No.** again. The number is stored, replacing any previous one.

To store a number during Ringback, while the number is busy, or during an active call (for Meridian 1 proprietary telephones only):

— Press **Stored No.**

To call a stored number:

1 Press **DN** (Meridian 1 proprietary telephones) or the Loop key (consoles).

2 Press **Stored No**. The number is dialed.

Analog (500/2500 type) telephones

To store a number prior to dialing:

1 Lift the handset.

2 Dial SPRE 78, or the Flexible Feature Code (FFC) assigned for SNR.

3 Dial the number to be stored.

4 Hang up. The number is stored, replacing any previous one.

To store a number before a call is placed, during Ringback, while the number is busy, or during an active call:

1 Flash the switchhook or press **LINK**.

2 Dial SPRE 78, or the FFC assigned for SNR.

To call a stored number:

1 Lift the handset.

2 Dial SPRE 79, or the FFC assigned for SNR. The number is dialed.

Introduced in X11 Release:	20.0x
Networking:	No

Telelink Mobility Switch 1

The Telelink Mobility Switch 1 feature allows a Meridian 1 in conjunction with the Mobility Control Point (MCP) application to deliver a call from the Public Switched Telephone Network (PSTN) to a portable telephone subscriber. The portable telephone subscriber does not need to have a telephone physically resident in the Meridian 1. A unique personal Directory Number not related to a physical termination is assigned for each subscriber of a portable. This unique personal directory number is defined as a Dialed Number Identification Services (DNIS) number within the Meridian 1 switch. Digit conversion is used to translate the incoming DNIS number to a Controlled DN (CDN). The CDN contains the Application Module Link (AML) number that connects the Meridian 1 to the MCP application.

When a call comes to the Meridian 1 (acting as the mobility switch) from the PSTN via a DNIS Incoming Digit Conversion (IDC) trunk, the call is terminated to a CDN by the Meridian 1 software through IDC operation. The last few dialed digits are saved as a DNIS (subscriber identity) number.

A CDN can be operated in controlled or default mode. If in controlled mode, call treatment is controlled by the MCP application. If in default mode, call treatment is handled by the Meridian 1 software and default treatment is given to the call.

When a Personal Communications Service (PCS) call terminates to a CDN which is in the controlled mode, the Meridian 1 will notify the MCP application by providing the call's incoming route and DNIS (subscriber identify) number. This enables the MCP application to ascertain which subscriber the caller desires to reach.

The MCP application has a table containing the last zone in which each subscriber is registered, so that the MCP application can send a message to the correct zone to find out the idle/busy status of the portable telephone subscriber.

If a subscriber is busy or unable to answer the call, the MCP application will request the Meridian 1 (acting as a mobility switch) to either return busy or overflow tone to the caller. If the called party has subscribed to a voice messaging service, the MCP can request the Meridian 1 (acting as a mobility switch) to allow the caller to leave a voice message. Meridian Mail can provide a busy tone or a no answer greeting to the caller (e.g., party X is busy, would you like to leave a message?)

If the called party is idle, the MCP application will request the Meridian 1 (acting as a mobility switch) to optimally give ringback, provide a Recorded Announcement (RAN) or give silence to the caller, while the MCP application requests the Meridian 1 to make an outgoing call that will be used to alert the called party that an incoming call is waiting. This outgoing call is initiated from a phantom TN. The phantom TN does not need a physical line connection or set in the Meridian 1. The phantom TN needs to be assigned as an associated set so that the MCP application will get status messages regarding the state of the phantom set. The public number of this outgoing call will be provided by a Zone Controller (ZC). The ZC reserves this incoming line which is connected to the public number.

When the phantom call is received on the reserved line, the ZC alerts the called party's portable. If MCP has requested silence for the call, then at this time the MCP will request ringback treatment for the call. Once the phantom call is answered by the subscriber, the ZC notifies the MCP application. Subsequently, the MCP application requests the Meridian 1 to merge the two related calls (an incoming call in the CDN queue and an outgoing call to the called party), so that the caller of the incoming call and the called party can speak to each other.

When an incoming call terminates to a CDN that is in default mode, the Meridian 1 (acting as a mobility switch) allows the caller to leave a voice message for the called party or give overflow tone to the caller when the call ceiling is exceeded. The CDN will be in default mode under abnormal conditions such as the AML, MCP or Application Programmable Interface (API) going down.

The Meridian 1 Mobility Switch will also provide centralized voice prompts in lieu of zones if an exception condition is encountered when a portable is attempting to make an outgoing call.

Operating parameters

Option 11 is not supported due to the phantom TN loop capacity.

There will be a 3 DB loss on a DTI trunk when a Digital Trunk Interface (DTI) trunk is involved in a merge call, and a 0 DB loss on a Primary Rate Interface (PRI) trunk when a PRI trunk is involved in a merge call. It is therefore recommended that a PRI trunk should be used on the Mobility Switch instead of a DTI trunk.

Calls to subscribers without physical sets on the Meridian 1 must be originated from DNIS routes.

The MCP application should request Force Overflow to an incoming call when the DNIS information is not present in an AML-ICC message.

This feature is supported for North American markets only.

ACD-C or ACD-D reports are not a requirement of this feature. Operational measurements of PCS calls are supported by the MCP application.

External calls coming to a CDN with Value Added Server Identification (VASID) connected to the MCP application from a DID (Digital or ISDN) trunk will only be supported by this feature. For this reason disconnect supervision will be guaranteed to be returned to the Meridian 1 (acting as a mobility switch) when the call is disconnected.

This feature only supports TIE, CO ground start (analog, digital or ISDN) trunks as the outgoing trunk of a phantom call with ZC as the destination. For this reason, disconnect supervision must be obtained from the far end when the call is disconnected by the far end.

Combination of CDNs and ACD-DNs (Interactive Voice Response-DN) cannot exceed 240 per each customer on the Meridian 1 (acting as a mobility switch).

The maximum number of customers per Meridian 1 Options 51, 61 or 71 systems cannot exceed 100.

The maximum number of routes cannot exceed 512.

The maximum number of IDC/New Flexible Code Restriction (NFCR) Translation Tables per customer cannot exceed 255.

Due to the Federal Communications Commission (FCC) ruling, answer supervision is required to be returned if a “Give Silence” or “Give Music” is provided as a first call treatment to a PCS (incoming DID) call as per current operation.

If the Meridian 1 (acting as a mobility switch) initializes, all calls waiting in the CDN queues will be lost. The AML-INIT message will be sent to applications when an initialization occurs. When the MCP application receives an INIT message from the Meridian 1 (acting as a mobility switch), it erases information on existing calls.

A maximum of five Device Groups (DGRPs) will be supported per customer. An Associated Set (AST) Meridian 1 proprietary telephone with Idle Terminal for Third Party Application (ITNA) enabled can only be grouped to one DGRP.

The originator of an outgoing phantom call must be a phantom TN which is an AST Meridian 1 proprietary telephone with ITNA enabled.

An attendant set and a Basic Rate Interface (BRI) set will not be allowed to merge a call to another set or trunk.

When two trunks are joined, at least one trunk must have disconnect supervision.

The Application Module (AM Base) that interfaces with the MCP application cannot control more than one application (i.e., MCP and Customer Controlled Routing applications are not supported).

A call that is initiated from the phantom set must be in established state, before it can be merged with the caller. If answer supervision is defined for the outgoing trunk, the call from the phantom set will be put in established state when the answer supervision answer is returned to the Meridian 1 (acting as a mobility switch). If answer supervision is not defined for the outgoing trunk, the call from the phantom set will be put in established state when the End-of-dialing timer has expired (128-32,640 msecs. after the last digit has been sent out).

If answer supervision is not defined for the outgoing trunk to which the phantom trunk is connected, it is possible that a random call may beat the phantom call to the reserved line and the caller will be given a busy tone.

If answer supervision is defined for the outgoing trunk, it is possible that the PSTN might not return the answer supervision signal to the Meridian 1. If answer supervision is not returned, the Meridian 1 will not allow the phantom set call to be merged to the caller.

If answer supervision is defined for the outgoing trunk, it is possible that the answer supervision signal could be significantly delayed across the PSTN if the signal goes through many tandem Central Offices. This causes a subsequent delay between the time the subscriber answers the portable and the time when the incoming call is connected.

No Message Waiting Indication will be sent to the MCP application when a caller has left a voice message.

The MCP application will not know if there is an invalid mailbox (Treatment DN) used to connect to Meridian Mail.

The MCP application must return a dialable number to the Meridian 1 (acting as a mobility switch) to launch the outgoing call to the Zone Controller. The dialable number includes the ESN access code if necessary.

If 1+ dialing is required at the first Central Office that the phantom call goes to from the Meridian 1, it should either be provided by the MCP application or inserted via digit manipulation on the mobility switch.

Enhanced Serial Data Interface (ESDI) (QPC 513 vintage G or later) or Multi-purpose Serial Data Link (MSDL) (NT6D80AA) is required to connect the Meridian 1 to the AM or a host.

Feature interactions

The following features interact with the Telelink Mobility Switch 1 feature:

— Report Control

— Print CDN Parameters and Options Command

— CNTL Command (determines whether CDN is in controlled mode)

— DFDN Command (sets default of ACD-DN)

— CEIL Command (controls ceiling of the CDN)

— Supervisor Control of Queue Size

— Overflow by Count

— Attendant Extension

— Attendant Recall

— Network ACD (NACD)

— Timed Overflow and Enhanced Overflow

— Display Waiting Calls (DWC key)

— Night Service

— Transition Mode via the Night Service key

— Night Mode via the Night Service key

— Incoming Digit Conversion

— Night Key Digit Manipulation

— Call Forward No Answer

— Call Forward No Answer (Second Level)

— Call Forward All Calls

— Internal Call Forward

— Feature Invocation Messages

— Hunting

— Call Forward Busy

— Remote Call Forward

— Attendant and Network Wide Remote Call Forward

— Network Call Redirection

— Call Forward Override

— Trunk Optimization

— Call Transfer

— Call Transfer – By Interactive Voice Response Unit

— Conference

— Conference – By Interactive Voice Response Unit

— No Hold Conference

— Calling Line Identification

— Basic Rate Interface

— Incremental Software Management

— PBS Set Line Disconnect

— Application Module Link Enhancements

— NCOS Restrictions

— Time-of-day Routing

— Expensive Route Warning Tone

— Off-hook Queuing

— Call Back Queuing

— Remote Virtual Queuing

— Authcode Last

— Equal Access

— 1+ Dialing

— Interchangeable Numbering Plan Area

— Inter Digit Pretranslation

— Free Call Area Screening

— New Flexible Code Restriction

— Call Forward on DumpSysload

— Flexible Numbering Plan

— Multi-party Operation

— Priority Override

— Group Hunt

— Virtual Network Services

— Originator Routing Control, and

— Enhanced Night Service.

The following are a list of features that interact with the Merge Call aspect of this feature:

— Tenant-to-tenant Access

— Class of Service Restrictions

— Network Class of Service (NCOS) Restrictions

— Trunk Group Access Restrictions

— Schedule Access Restrictions

— Trunk Barring

— Feature Group D

— Attendant Barge-in

— Attendant Break-in

— Attendant Busy Verify

— Transfer

— Conference

— No Hold Conference

— Call Waiting

— Internal Call Waiting

— Group Call

— Voice Call

— Call Park
— Station Camp-on
— Dial Intercom
— ACD-DN key
— ACD Emergency/Answer Emergency keys
— ACD Call Agent/Answer Supervisor keys
— ACD Summon Supervisor/Answer Agent keys
— Single Call Arrangement DN keys
— Multiple Call Arrangement DN keys
— HOT Line
— Private Line
— Integrated Service Access Routes
— Integrated Signaling Link
— Application Module Link Unsolicited Status Message, and
— Application Module Link Call Abandoned Message, and Digit Display.

Feature packaging

There is no new software package required for this feature; however, the following existing packages are required for it to operate:

— Automatic Call Distribution, Package B (ACDB) package 41
— Network Alternate Route Selection (NARS) package 58
— Command Status Link (CSL) package 77
— Dialed Number Identification Services (DNIS) package 98
— Incoming DID Digit Conversion (IDC) package 113
— Application Module Link (AML) package 153
— Meridian Link Module (MLM) package 209
— Enhanced ACD Routing (EAR) package 214
— Enhanced Call Treatment (EACT) package 215

— Hold in Queue for Interactive Voice Response (IVR) package 218

— Call Identification (CID) package 247

— Phantom Terminal Number Operation (PHTN) package 254

The following packages are not required, but provide additional functionality:

— Call Detail Recording (CDR) package 4

— Office Data Administration System (ODAS) package 20

— ACD Load Management (LMAN) package 43

— Multi-user Login (MULI) package 242

Feature implementation

LD 11 – Two new prompts have been introduced to this overlay: ITNA and DGRP.

Prompt	Response	Description
REQ	NEW CHG MOV OUT END CHG	New, change, move, out, end, or change.
TYPE	SL1 2006 2008 2009 2016 2018 2112	Type of Meridian 1 proprietary telephone.
CUST	0-99	Customer number.
TN	l s c u	Terminal Number. If l is a phantom loop and the CSL package is not equipped, an error message will be returned.
TOTN	l s c u	Loop, shelf, card, and unit (destination TN). Prompted when REQ = MOV.
CFTN	l s c u	Loop, shelf, card, and unit (destination TN). Prompted when REQ = CPY.
SFMT	AUTO DN TN TNDN	DNs and TNs are assigned automatically. User enters the DN for each new telephone. User enters the TN for each new telephone. User enters DN and TN for each new telephone. Prompted when REQ = CPY.

CDEN	YES	Single, double or quad density (not prompted for superloop).
DES		ODAS designator.
...		
CLS	 (NDD) (DNDD) ...	Class of service options. No digit displayed. Dialed name display denied. Block SPV and AGN if this TN is on a phantom loop.
AST	xx yy	Associate telephone assignment for Meridian Link application.
IAPG	(0)-15	Meridian Link Unsolicited Status Message (USM) group.
ITNA	(NO) YES	Idle TN for the third party application.
DGRP	(1)-5	Device group
...		
KEY	xx SCR yyyy	Single call ringing DN key.

LD 17 – Configure a phantom loop.

Prompt	Response	Description
REQ	CHG END	Change, or end.
TYPE	CFN CEQU	Configuration Record. Release 19 gate opener.
CEQU	YES	Change Common Equipment parameters. This will be prompted if TYPE = CFN.
...		
- TERM	0-159 0-159... [X] 0-159 [C] 0-159...	Single density local terminal loops. Precede loop number with X to remove. Precede the loop number with C to create a phantom loop.
- REMO	0-159 0-159... [X] 0-159	Single density remote terminal loops. Precede loop number with X to remove.

- TERD	0-159 0-159... [X] 0-159 [C] 0-159...	Double density local terminal loops. Precede loop number with X to remove. Precede the loop number with C to create a phantom loop.
- REMD	0-159 0-159... [X] 0-159	Double density terminal loops.
- TERQ	0-159 0-159... [X] 0-159 [C] 0-159...	Quad density local terminal loops. Precede loop number with X to remove. Precede the loop number with C to create a phantom loop.
- REMQ	0-159 0-159... [X] 0-159	Quad density remote terminal loops. Precede loop number with X to remove.

LD 97 – Create a phantom loop (superloop).

Prompt	Response	Description
REQ	CHG	Change.
TYPE	SUPL	Superloop data.
SUPL	0-156 [X] 0-156 [C] 0-156	Superloop number in multiples of four. Precede loop number with X to remove a superloop. Precede the loop number with C to create a phantom superloop.

LD 23 – This overlay is used to administer a service change to a CDN data block.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	CDN	Controlled DN data block.
CUST	0-99 0-31	Customer number. For Option 11C.
CDN	Directory Number	Controlled DN
FRRT	RAN route number	First RAN route number.

FRT	1-2044	First RAN timer.
SRRT	RAN route number	Second RAN route number.
SRT	1-2044	Second RAN route timer.
FROA	(NO) YES	First RAN to be given immediately.
MURT	Music route	Music route number.
DFDN	Directory Number	Local default ACD-DN or IVR DN.
CEIL	0-(2047)	Call ceiling value.
OVFL	(NO) YES	Force Overflow Tone to the call when ceiling threshold exceeded?
TDNS	(NO) YES	Is the DNIS number an original party?
RPRT	(YES), NO	Report on this CDN in ACD-C or D reports.
CNTL	(NO) YES	Is this CDN in controlled mode?
VSID	0-15	VASID for AML for application.
HSID	0-15	VASID for AML for host.
CWTH	0-(1)-2047	Call waiting LED threshold.
BYTH	(0)-2047	Busy queue threshold.
OVTH	0-(2047)	Overflow queue threshold.
STIO	1 2 3 ... 15	TTYs assigned for status displays.
TSFT	0-510	Telephone service factor threshold.
ACNT	xxxx	Default activity code.

Feature operation

No specific operating procedures are required to use this feature.

Telephones

There are several different types of telephones you can use in the Meridian 1 system. Regular telephones are compatible with the Meridian 1 system, as well as several special business telephones designed specifically to take advantage of the many features available.

This module provides an overview of the telephones and a description of the basic features and services. Additional information regarding related software features is found in other modules of this document.

"Meridian 1 proprietary telephones" is used as a generic term and includes the SL1, M2000 series telephones, the M2317 telephone, the M3000 Touchphone and Meridian Modular Telephones, which are all configured using LD 11.

Analog (500/2500 type) telephones

Analog (500/2500 type) telephones are regular telephones not normally associated with a business environment, but they are compatible with the Meridian 1 system. They are configured by using LD 10. The 500-type telephones have a rotary dial. The 2500-type telephones are the basic push-button models, such as the Link and Unity, which do not have feature buttons normally found on business telephones.

Although analog (500/2500 type) telephones do not have feature keys, you can access various system features using Special Prefix (SPRE) codes. SPRE codes are also useful for Meridian 1 proprietary telephones to access features without using feature keys. Dial the SPRE code (unique to each customer within the system) and then the feature code that applies to the operation you desire.

Table 140 lists the feature codes available using SPRE.

Table 140
Feature codes used with SPRE (Part 1 of 2)

Dial SPRE +	Operation performed
1	Ring Again
2	Cancel Ring Again
3	Ringing Number, Call Pickup
4	TAFAS (Trunk Answer From Any Station)
5	Charge Account for CDR
6	Authorization Code Access
70 + ACOD + mmm (Trunk Route Access Code and Member)	Trunk Verification From Station
71 + DN	Call Park, To Park
72 + DN	Call Park, To Retrieve
73	System Speed Call, To Use
74	Call Forward activate or cancel (500-type telephones)
75 + Entry Access Code + DN (500-type telephones)	Speed Call, Individual To Program Entry
76 + Entry Access Code (500-type telephones)	Speed Call, Individual To Use Entry
77	Permanent Hold (500-type telephones)
78	Stored Number Redial, To Store
79	Stored Number Redial, To Redial
81	Automatic Set Relocation
83	Malicious Call Trace
84	Integrated Messaging System
86 + x (status)	Room Status
86 + 1	Cleaning Request
86 + 2	Cleaning In Progress
86 + 3	Room Cleaned

Table 140
Feature codes used with SPRE (Part 2 of 2)

Dial SPRE +	Operation performed
86 + 4	Passed Inspection
86 + 5	Failed Inspection
86 + 6	Cleaning Skipped
86 + 7	Not For Sale
87	Disconnect Trunk, Conference 6 (analog (500/2500 type) telephones)
89	Last Number Redial
91	Access to maintenance programs by Maintenance Telephone
92	Terminal Diagnostics, telephones and Attendant Consoles
93	Conference Circuit Testing
94	Ringing Number, Group Pickup
95	Ringing Number, DN Pickup
96	Centrex Switchhook Flash
97	Unassigned Automatic Call Distribution (ACD) analog (500/2500 type) telephone Log in/out
98	Unassigned ACD analog (500/2500 type) telephone Activate/deactivate Not Ready

Table 141
2500-type telephone features (no SPRE code used)

# + 1 + DN	Call Forward
# + 2 + Speed Call code + DN	Speed Call, Individual, To Program Entry
# + 2 + Speed Call code + *	Speed Call, Individual, To Erase Entry
# + 3 + Speed Call code	Speed Call, Individual, To Use Entry
# + 4	Permanent Hold

SL-1 telephones

The SL-1 telephone is designed specifically for the Meridian 1 system and allows the user to access many system features. All SL-1 telephones are equipped with a 12-key dial pad, 10 feature keys and three fixed control keys. Table 142 summarizes the different models of SL-1 telephones.

Table 142
SL-1 telephones

Set type	Comments
QSU1	No display.
QSU3	Same as QSU1, with a 16-character display window.
QSU6	Same as QSU1, with two headset or handset jacks. Intended for Automatic Call Distribution (ACD) operations.
QSU7	Same as QSU3, with two headset or handset jacks. Intended for Automatic Call Distribution (ACD) operations.
QSU60	Similar to QSU1, with minor alterations for the U.S. market.
QSU61	Similar to QSU3, with minor alterations for the U.S. market.
QSU71	The Meridian M1109 telephone. Similar to the QSU1, with built-in Handsfree.

The SL-1 telephone is designed to accommodate various add-on modules to increase its functionality. Table 143 lists the modules you can add on to an SL-1 telephone.

Table 143
Add-on modules available for SL-1 telephones

Add-on module	Description	Comments
QMT1	10 key/lamp strip	Requires additional power
QMT2	20 key/lamp strip	Requires additional power
QMT3	Lamp Field Array	Requires additional power
QKK1	Handsfree Interface kit	Requires additional power
QKK3	Automatic Handsfree Interface kit	Requires additional power
QKK8	Answerback Interface kit	QSU71 only
QKN1	Headset interface kit	
QSAM3	Group listening switch	Allows caller to be heard through set's loudspeaker
QMT15	Amplified Handset	Requires Current Limiting Kit (P0630408)
QKM13	Light Probe Kit	For sight-impaired users

M2000 series digital telephones

M2000 series digital telephones are available on X11 Release 7 and later software. They are designed to provide integrated voice and data communication. Use the M2000 Asynchronous Data Option to make data calls. There are three models in the M2000 series:

— M2009 has nine programmable keys;

— M2018 has 18 programmable keys; and

— M2112 has 11 programmable keys and one fixed Handsfree key.

M2000 series digital telephones are not designed for use in an ACD environment.

M2317 digital telephones

The M2317 digital telephone is available on X11 Release 9 and later software. It is equipped with a two-line (40 characters per line) liquid crystal display (LCD) screen and integrated Handsfree. To make data calls, you need an M2000 Asynchronous Data Option.

Five soft, or screen-dependent, keys are located beneath the display screen. These keys, when operated, activate the function that the screen describes as being accessible. Each softkey is associated with a label, seven characters wide, on the display screen immediately above the key.

Softkeys are designated as key numbers 17 through 29. When the M2317 is configured in the system software, certain default features are automatically assigned to the softkeys. Some features cannot be added to the softkeys. See Table 144 for a description of softkey feature assignments.

Key 11 automatically defaults to Handsfree and cannot be assigned. Keys 12 through 16 and key 18 are reserved for future development and cannot be assigned. The second appearance of a data DN must be assigned to key 10 on the voice TN for keypad dialing.

M3000 Touchphone

The M3000 Touchphone is available on X11 release 7 and later software. It is a digital, integrated voice/data telephone with a touch-sensitive liquid crystal display (LCD) screen and integrated Handsfree. An M3000 Asynchronous Data Option provides data call capability.

Table 144
M2317 softkey feature assignments

Key No.	Mnemonic	Feature
Default feature assignments:		
11		Handsfree/mute
17	PRK	Call Park
23	AO6	Conference 6
24	CPN	Calling Party Number
25	CHG	Charge Account
26	TRN	Call Transfer
27	RGA	Ring Again
28	PRS	Privacy Release
29	LNG	Language
Keys reserved for specific features (programmed in LD 11):		
19	RNP	Ringing Number Pickup
20	MWK	Message Waiting
21	SSU, SSC SCU, SCC	Speed Call or System Speed Call
22	CFW	Call Forward
Note: Default key assignments are activated only if the feature is part of your software package, the feature is defined for this customer, and the feature is allowed for the telephone.		

All features are displayed on the screen and are accessed by touching the appropriate name on the screen. The M3000 can display a number of online feature descriptions and operating instructions in user-friendly language.

The M3000 has a directory that can store from 150 to 450 numbers (up to 28 digits) and names (up to 15 characters) that you can access by simply touching the screen. You can search the directory or scroll the display up or down, and dial the desired telephone number by touching the name on the screen.

The M3000 Touchphone is not designed for use in an ACD environment.

When the M3000 is configured in the system software, certain default features are automatically assigned to the telephone. Table 145 gives information on feature key assignments.

The second appearance of a data DN must be assigned to key 17 on the voice TN for keypad dialing.

Table 145
M3000 feature key assignments (Part 1 of 3)

Key No.	Mnemonic	Feature
0-5	SCR	Single Call Ringing
	MCR	Multiple Call Ringing
	DIG	Dial Intercom Group
	PVR	Private Line Ringing
	COS	Controlled Class of Service
6-16		Reserved for future development

Table 145
M3000 feature key assignments (Part 2 of 3)

Key No.	Mnemonic	Feature
17	SCR	Second appearance of data DN (if CLS = DTA)
18	SIG	Manual Signaling (Buzz)
19		Reserved for future development
20	MWK	Message Waiting
21	SCU SCC SSU SSC	Speed Call User Speed Call Controller System Speed Call User System Speed Call Controller
22	CFW	Call Forward All Calls
M3000 default feature assignments:		
23	AO6	Conference 6
24	CWT	Call Waiting
25	CHG	Charge Account
26	TRN	Call Transfer
27	RGA	Ring Again
28	PRS	Privacy Release
29		Reserved for future development
30	MSB	Make Set Busy
31	PRK	Call Park
32	CPN	Calling Party Number
33	ARC	Attendant Recall
34	OVR	Override
35	AAK	Automatic Answerback
36	DSP	Display

Table 145
M3000 feature key assignments (Part 3 of 3)

Key No.	Mnemonic	Feature
Features not supported by the M3000:		
	NHC	No Hold Conference
	CS	Combined No Hold Conference and Speed Call
	DPU	Directed Call Pickup
	GRC	Group Call
	GPU	Group Number Pickup
	VCC	Voice Call
Note: Default key assignments are activated only if the feature is part of your software package, the feature is defined per customer, and the feature is allowed in Class of Service.		

Meridian Modular Telephones

The Meridian Modular Telephones are available with X11 Release 14 and later software. They are designed to provide cost-effective integrated voice and data communication capability. These telephones communicate with the Meridian 1 and SL-100, using digital transmission over standard twisted-pair wiring. Table 146 summarizes the different models of Meridian Modular Telephones.

When a modular telephone is equipped with either a display or data option, a PROGRAM key (key 5 for M2006, key 7 for all remaining modular telephones) is automatically assigned to the upper right-hand feature key. This feature provides user control over such display features as screen format, contrast and language. It also provides user control over such parameters as transmission speed, parity and terminal mode.

Table 146
Meridian Modular Telephones

Set type	Programmable keys	Additional comments
M2006	6	Single-line only

Table 146
Meridian Modular Telephones

Set type	Programmable keys	Additional comments
M2008	8	Multi-line
M2616	16	Programmable Handsfree
M2016S	16	Telephone Security Group Class II approved
M2216ACD-1	16	ACD Display module and two RJ-32 headset jacks
M2216ACD-2	16	ACD Display module; one RJ-32 and one PJ-327 headset jack

The Meridian Modular Telephones are designed to accommodate various add-on modules to increase their functionality. Table 147 lists the modules you can add on to a Meridian Modular Telephone.

Table 147
Add-on modules for Meridian Modular Telephones

	M2006	M2008	M2016S	M2616	M2216ACD-1	M2216ACD-2
Display		x	x	x	Standard	Standard
Key Expansion Module			x	x	x	x
Programmable Data Adapter	x	x	x	x	x	x
External alerter interface	x	x		x	x	x
Note: In this table, x indicates available add-ons for the telephone listed along the top row.						

M2006

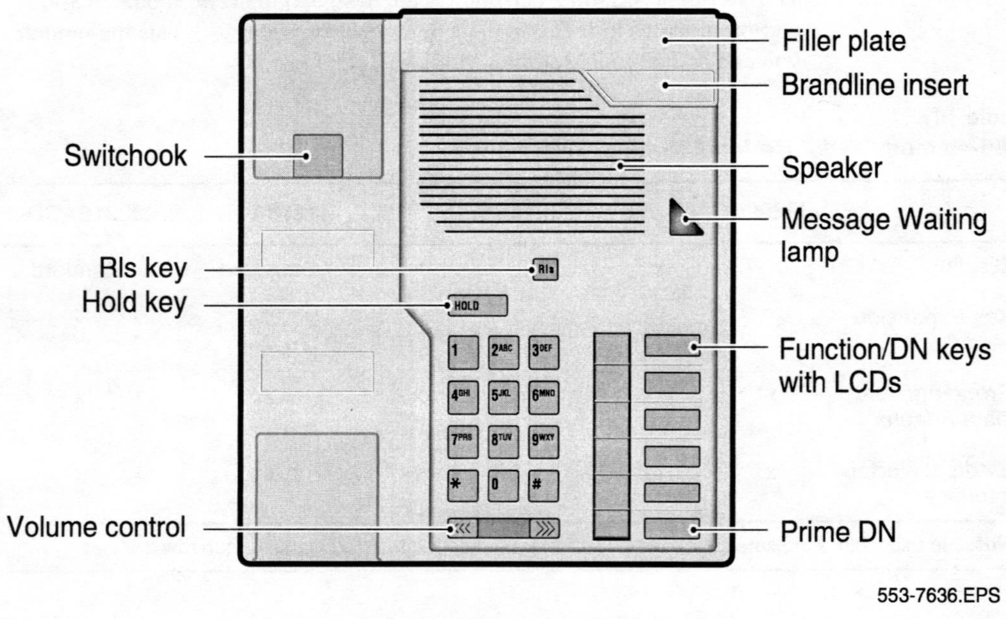
The M2006 is a digital single-line telephone that provides on-hook dialing, volume control, Release and Hold keys and a Message Waiting indicator. In addition, it provides four or five programmable feature keys (five if the data option is not in use). It also has a one-way speaker and a programmable data option.

The M2006 can have an optional external alerter interface, which connects to any standard remote alerting device.

The M2006 works off any digital line card.

Figure 86 shows the M2006 telephone.

Figure 86
M2006 telephone



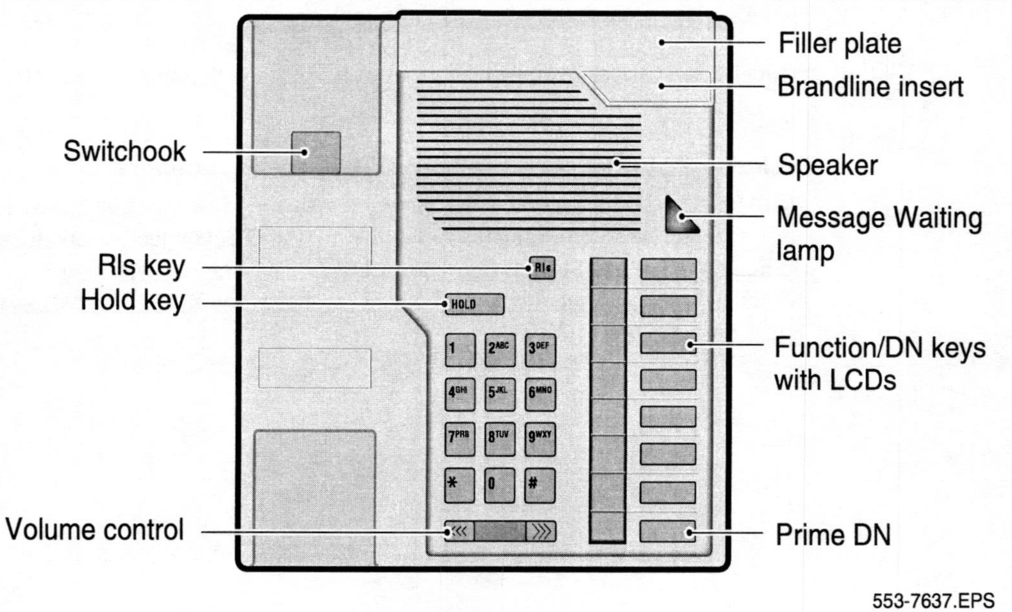
M2008

The M2008 digital telephone has eight programmable feature/line keys, on-hook dialing, volume control, Release and Hold keys and a Message Waiting indicator.

The M2008 also supports the programmable data adapter, alphanumeric display and external alerter interface options.

Figure 87 shows the M2008 telephone.

Figure 87
M2008 telephone



M2616, M2216 (Models 1 and 2)

The M2616 telephone has 16 programmable feature/line keys, on-hook dialing, volume control, Release and Hold keys, Message Waiting indicator and Handsfree/mute features. It supports up to two add-on modules (each of 22 keys), an alphanumeric display option (two lines of 24 characters each), programmable data adapter and an external alerter interface.

The M2216 Model 1 and the M2216 Model 2 are almost identical to the M2616 with the following exceptions:

— They have no switchhook because they are designed for plug-in handset or headset operation.

— Display is standard rather than optional.

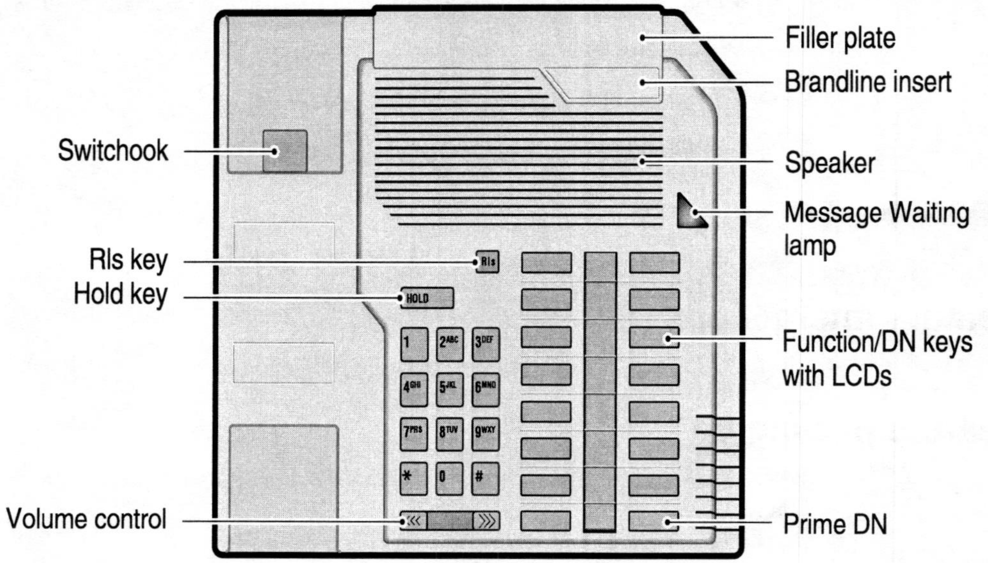
Model 1 and Model 2 refer to the types of headsets with which the M2216 operates.

Release 22 Enhancements to M2216 Voice Parameters

With Release 22, the receive voice parameters on the M2216 can be increased up to 6 decibels. To program this capability, the ACD Set Objective Loudness Rating (AOLR) must be configured in LD 17. See "LD 17 – Meridian Modular Telephones related prompts and responses for X11 Release 22 and later" on page 2750.

Figure 88 shows the M2616 telephone.

Figure 88
M2616 telephone



553-7635.EPS

Related documentation

Refer to the following publications for additional information on telephones and add-on modules.

— *Meridian 1 telephones description and specifications* (553-3001-108);

— *Digital telephone line engineering* (553-2201-180);

— *Telephone and Attendant Console Installation (553-3001-215);* and

— X11 input/output guide.

Operating parameters

Refer to the preceding Northern Telecom publications.

Feature interactions

Refer to the preceding Northern Telecom publications.

Feature packaging

Analog (500/2500 type) telephone and SL-1 telephone capabilities are included in base X11 system software.

Digital Sets (DSET) package 88 has no feature package dependencies (Meridian M2000 series telephones).

M2317 telephone (DLT2) package 91 requires Digital Sets (DSET) package 88.

M3000 Touchphone (TSET) package 89 requires Digital Sets (DSET) package 88.

Meridian Modular Telephones (ARIE) package 170 requires Digital Sets (DSET) package 88, and M3000 Touchphone (TSET) package 89.

Feature implementation

LD 15 – Add, or change an existing Special Prefix Code (SPRE).

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options.
CUST	0-99 0-31	Customer Number. For Option 11C.
SPRE	xxxx	Special Prefix number. The prefix must not conflict with the numbering plan.

LD 10 – Add or change analog (500/2500 type) telephone.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
CDEN	SD (DD) 4D	Card density (single, double, quad). This prompt appears only if no units on the card have been defined.
DES	a...x	Set designator (1-6 characters, alphanumeric).
CUST	xx	Customer number.
DN	xxx...x	Directory number.
TGAR	0-xx	Trunk Group Access Restriction.
CLS	aaa	Class of Service mnemonics for feature assignment.

LD 11 – Add or change Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
CDEN	SD DD 4D	Card density (single, double, quad). Not prompted for octal density. This prompt appears only if no units on the card have been defined. Card density must be 4D if TYPE is not SL-1.
DES	a...x	Designator (1-6 characters, alphanumeric).
CUST	xx	Customer number.
AOM	(0)-2	Number of key expansion modules. Prompted if TYPE = 2016, 2216, or 2616.
KLS	1-7	Number of key/lamp strips (SL-1 telephones only).
TGAR	0-xx	Trunk Group Access Restriction.
CLS	aaa	Class of Service mnemonics for feature availability.
KEY	xx aaa yyy...y	DN and feature key assignment (key number, feature mnemonic, directory number if applicable).

Note 1: A Message Waiting Allowed (MWA) Class of Service must be defined to enable the message waiting lamp.

Note 2: Key 7 (key 5 for M2006) is reserved for the PROGRAM key (M2008, M2016S, M2216ACD, M2616) only if display or data is equipped.

LD 17 – Meridian Modular Telephones related prompts and responses, Release 18 and later.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATRN	Release 19 gate opener.
ATRN	(NO) YES	Change transmission parameters.
- CODE	(0)-2	CODEC Coding Law.
- SOLR	0-(1)-4	Sidetone Objective Loudness Rating.
- ROLR	(0)-63	Receive Objective Loudness Rating.
- AOLR	(0)-12 32-50	2216 ACD Set Objective Loudness Rating.
- TOLR	(0)-63	Transmit Objective Loudness Rating.
- AGCD	(NO) YES	Automatic Gain Control disabled.
Note: Default settings are recommended. See *Summary of transmission parameters* (553-2201-182) before changing these parameters.		

LD 17 – Meridian Modular Telephones related prompts and responses for X11 Release 22 and later

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATRN	Release 19 gate opener.
ATRN	(NO) YES	Change transmission parameters.
- CODE	(0)-2	CODEC coding law.
- SOLR	0-(1)-4	Sidetone Objective Loudness Rating.
- ROLR	(0)-63	Receive Objective Loudness Rating.
- AOLR	(0)-12 32-50	2216 ACD Set Receive Objective Loudness Rating.
- TOLR	(0)-63	Transmit Objective Loudness Rating.
- AGCD	(NO), YES	Automatic Gain Control disabled.
Note: Default settings are recommended. See *Summary of transmission parameters* (553-2201-182) before changing these parameters.		

LD 11 – Add data TN to Meridian Modular telephones.

Prompt	Response	Description
REQ:	NEW	New.
TYPE:	aaaa	Telephone type, where: aaaa = 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
DES	a...x	Set designator (1-6 characters alphanumeric).
CLS	aaa	Class of Service mnemonics for feature availability.
DTYP	IOS	Inbound/outbound data station.

ADCP	(NO) YES	All digital connection prefix.
KEY	xx aaa yyy...y	DN and feature key assignment (key number, feature mnemonic, directory number if applicable). ***Note:*** Recommended key assignments for data TN are: Key 0 = DN (for data) Key 1 = DN (secondary) Key 2 = TRN (Transfer) Key 3 = ADL xxxx (Auto Dial Directory Number) Key 4 = RGA (Ring Again) Key 5 = SSC, Sc u, SSC, SSU (Speed Call, System Speed Call, controller or use – not available on M2006), and Key 6 = DSP (Display key for M2008, M2016S, M2216ACD, M2616).

LD 11 – Add data TN to SL-1 telephones with data module.

Prompt	Response	Description
REQ:	NEW	New.
TYPE:	SL1	SL-1 telephone.
TN	l s c u c u	TN location (loop, shelf, card, unit). Unit number equals the voice TN unit number plus eight. For option 11C.
CUST	xx	Customer number.
CLS	WTD	Warning Tone Denied.
KEY	xx aaa yyy...y	DN and feature key assignment (key number, feature mnemonic, directory number if applicable). ***Note:*** Recommended key assignments for data TN are: Key 0 = DN (for data) Key 1 = DN (secondary) Key 2 = TRN (Transfer) Key 3 = ADL xxxx (Auto Dial Directory Number) Key 4 = RGA (Ring Again) Key 6 = SSC, SSU (Speed Call controller or user), and Key 9 = Rls (Release).

Feature operation

Refer to the appropriate Telephone User Guide for more information on how to operate your telephone sets.

Introduced in X11 Release:	18.20H
Networking:	No

Teletype Terminal Access Control in Multi-customer Environment

This is an enhancement of password usage for the Limited Access to Overlays feature. Under the previously enhanced operation, if no teletype terminal (TTY) activity had occurred for 20 minutes, the system automatically logged off. This value could not be changed. Counters were used to record the number of login attempts made on each TTY. If the threshold for the number of invalid attempts was exceeded, the system rejected any further activity at that port, for a defined period of time. No alarm mechanism was activated. Any attempt to log into the system during this period of lockout was recorded by the system.

A new prompt (LOUT) has been introduced in LD 17 to allow the TTY administrator (PWD2 user) to define a period of time (1-20 minutes) after which the system automatically logs out if no terminal activity has occurred.

The recording of invalid attempts remains the same as before. However, if the threshold for the number of invalid entries is reached, an alarm is activated; this alarm is in the form of the "minor alarm" lamp being lit on Attendant Consoles for all customers of the system. As was the case for the previously enhanced operation, an OVL400 message is sent to all active maintenance ports and to the first TTY administrator that logs in. Other treatments also remain the same.

Operating parameters

There are no operating parameters.

Feature interactions

Intercept Computer

The Intercept Computer (ICP) feature uses maintenance LD 51 to update the Meridian 1 with the intercept service interface information that it stored. This overlay logs off after five minutes if no messages have been received from the Intercept Computer. This five-minute period takes precedence over the value entered in response to the LOUT prompt in LD 17. If this value is less than five minutes, the system will wait for five minutes before logging off.

Feature packaging

International Supplementary Features (SUPP) package 131; Limited Access to Overlays (LAPW) package 164.

Feature implementation

LD 17 – Configure TTY Access Control parameters.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PWD	Configuration Record. Release 19 gate opener.
- NPW1	xxx	New Password 1
- LOUT	1-(20)	Enter the time, in minutes, after which the system logs off if no terminal activity is detected.
...		
- FLTH	(0)-7	Enter the threshold for failed log-in attempts.
- LOCK	0-(60)-270	Enter, in minutes, the time the port is locked out once the FLTH has been reached.
- FLTA	(NO) YES	Enter YES to have the alarm activated once FLTH has been reached.
- AUDT	(NO) YES	Enter YES to have an audit trail activated for password usage.

- - SIZE	(50)-100	Prompted if AUDT = YES. Enter the size of the audit trail buffer.
- LLID	(NO) YES	Enter YES to activate the display of the last failed log-in attempt usage.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	15.58F
Networking:	No

Telset Call Timer Enhancement

The M2317, M3000, and new Meridian digital telephones have displayable call timers, which start after the End-of-dialing (EOD) time out expires, and not when the called party answers. With this enhancement, the call timers on these telephones do not start until a true answer is detected on all trunks with answer supervision. These include the following:

— internal stations and attendants

— ground start and loop start supervisory trunks

— Direct Inward Dialing (DID) and Direct Outward Dialing (DOD) trunks

— Digital Trunk Interface (DTI) trunks

— Primary Rate Interface (PRI) trunks, and

— TIE trunks.

On trunks without answer supervision, the call timer starts at the EOD time out.

The feature operates in standalone or Integrated Services Digital Network (ISDN) environments.

Operating parameters

There are no feature requirements.

Feature interactions

There are no interactions with other features.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

No change to existing configuration is required for the Telset Call Timer Enhancement feature.

Feature operation

No specific operating procedures are required to use this feature.

Three-Wire Analog Trunk – Commonwealth of Independent States

The Three Wire Analog Trunk – Commonwealth of Independent States (CIS) feature provides the connectivity between the Meridian 1 and the three-wire analog trunks (3WT) used in the CIS. Analog incoming local three-wire trunks, analog incoming toll three-wire trunks, and analog outgoing Direct Inward Dialing (DID) three-wire trunks can be connected to the Meridian 1.

The following hardware cards are supported:

— Cards supported in an Enhanced Peripheral Equipment (EPE) environment are referred to as E3W cards. They consist of:

- QPC661 for incoming trunk calls.
- QPC661 for incoming toll calls.
- QPC661 for outgoing 3WT local trunks.

— Cards supported in an Intelligent Peripheral Equipment (IPE) environment are referred to as X3W cards. They consist of:

- NT5K60AA for incoming local and toll trunks
- NT5K61AA for outgoing trunks.

The following functions are provided by the Three-Wire Analog Trunk – CIS feature:

— Delivery of Automatic Number Identification (ANI) on request from the Public Exchange/Central Office for outgoing 3WT analog calls

— Downloading of specific transmission parameters (i.e., pad data, public network toll access code, and hardware ID) for X3W cards, and

— Provision of dial tone internally by the Meridian 1 to the originator of the call after seizure of an outgoing X3W trunk.

The trunk state change validation timing is performed by the 3WT cards. For 3WT trunks, the originating party controls the disconnection of a call. When the originating party goes on-hook, the call is released. Note however, that when Malicious Call Trace is enabled, the Local Exchange may require a two-way release. This two-way release applies only on a set.

A 3WT Unproductive Timer is used to prevent a call on a X3W trunk from remaining unanswered for too long. This timer can be set to a maximum of 10 minutes.

For outgoing calls, digits are sent from the main Central Processing Unit (CPU) to the 3WT firmware. This is done by Dual-tone Multifrequency (DTMF) signaling for E3W equipment, and by IPE messaging for X3W equipment. The firmware then sends the digits as pulses and controls the actual decadic outpulsing.

Digits for incoming calls are received by the 3WT firmware as pulses. For E3W equipment, each valid pulse is reported to the main CPU by Scan and Signaling Distributor (SSD) messages. For X3W equipment, the pulses are collected by firmware and complete digits are reported to the main CPU as IPE digit messages.

Operating parameters

X3W trunk cards can only be configured on IPE shelves; E3W trunk cards can only be configured on EPE shelves.

Trunk-to-trunk connections are supported, but the ANI information will refer to the ANI DN of the incoming route, except with QSIG, Q931, and Digital Private Signaling System #1 (DPNSS1) routes. QSIG, and Q931 ANI information will use the Calling Line Identification (CLID) information, whereas DPNSS1 ANI will use the Originating Line Identifier (OLI) information if this information is present.

The Dynamic Loss Switching feature is not supported, because there is no connection matrix and loss alternative table available for the CIS market. However, Dynamic Loss Switching is supported in Australia, New Zealand, Italy, and China.

The Static Loss Plan Download (SLPD) feature is supported on X3W trunks.

No loss downloading/switching is done for E3W trunks.

ANI is only supported for outgoing calls.

The data in ANI is built only once at the beginning of the call. Once the trunk access code is dialed, the ANI information is downloaded to the 3WT firmware. The download of ANI occurs only once and is not changed or redownloaded for any kind of operation during a call; therefore, if the call goes through any type of modification such as a transfer or call forward for instance, the ANI information sent when requested is that of the original originator of the call.

Toll Operator Manual Ringing and Break-In are not supported on IPE analog trunks.

Data calls are supported, but with the limitations due to the 500 Hz ANI requests that can happen any time during the call and the ANI information being sent on the same voice circuit on which the data is being transmitted; therefore, the transmission of data is not guaranteed.

Multifrequency Shuttle signaling is not supported on either X3W or E3W trunk cards.

EPE interfaces cannot be used on the Option 11.

The CIS A-law XCT (NTD17AE) is required.

Feature interactions

Authorization Code

An extension may, referring to the Authorization Code, seize an outgoing CIS 3WT trunk. The Authorization Code category is used to build the ANI message, meaning that a set which has a CIS restricting call category can complete a call to the public network using the Authorization Code.

Autodial

Autodial on a E3W trunk will fail for toll calls. The reason is that E3W trunks do not wait for the ANI request from the Public Exchange/Central Office, which is expected to appear after the toll access code is dialed. The Public Exchange then does not accept the call due to failure to receive ANI information.

Dial Tone Detection

Dial Tone detectors are supported with the limitations of the reliability of the tone provided by the Public Exchange.

DPNSS1 Gateway

The ANI information transmitted for this incoming DPNSS1 route will include the Local Exchange Code (LEC) of the CIS outgoing route, the ANI DN, and the Category Code (CAC) of this incoming route.

The ANI DN information which is built will refer to the Originating Line Identifier (OLI) if present and the Route DN Length prompt for ANI (RDNL ¦ 0) in LD 16. If the OLI is available, but RDNL = 0 for that route, the ANI DN is the ANI DN of that incoming route. If the OLI is available, but RDNL = 0 and the ANI DN of the incoming route is not defined, the ANI DN is the ANI DN of the CIS outgoing route. If the OLI is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, and the ANI DN of the CIS outgoing route is not defined, the ANI DN will be built with the Additional Digit (ADDG). If RDNL ¦ 0, its value will be the number of digits extracted from the OLI to be used as the ANI DN. The least significant digit of the OLI will be extracted (for example, if the DN is 4201, the 1 is the least significant digit.)

If there is no OLI, the ANI DN of the DPNSS1 route is used to build the ANI message. If there is no ANI DN on the DPNSS1 route, the ANI DN of the CIS outgoing route is used to build the ANI message. If there is no ANI DN on the CIS outgoing route, the ANI is built with the ADDGs of the CIS route (ADDG is always defined).

Incoming Digit Conversion Enhancement
Incoming DID Digit Conversion

The construction of an ANI message does not care if Incoming Digit Conversion is used. The DN sent as ANI is the actual DN of the set, not necessarily the DID number to dial to reach the set. Therefore, if an external party uses a DN for making a call to the corresponding extension which is delivered in an ANI message, the call may fail.

Last Number Redial

Last Number Redial on an E3W trunk will fail for toll calls. The reason is that E3W trunks do not wait for the ANI request from the Public Exchange, that is expected to appear after the toll access code is dialed. The Public Exchange will not accept the call due to the failure to receive ANI information.

Multiple Appearance Directory Number

Since the ANI category is defined on a per set basis for Three Wire Analog Trunks, two stations with the same multiple Appearance DN can be assigned different ANI categories

Q931 Gateway/BRI Gateway

The ANI information transmitted for this incoming Q931 route will include the LEC of the CIS outgoing route, the ANI DN, and the CAC of this incoming route.

The ANI DN information which is built will refer to the Calling Line Identification (CLID) if present and the Route DN Length prompt for ANI (RDNL ¦ 0) in LD 16. If the CLID is available but RDNL = 0 for that route, the ANI DN is the ANI DN of that incoming route. If the CLID is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, the ANI DN is the ANI DN of the CIS outgoing route. If the CLID is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, and the ANI DN of the CIS outgoing route is not defined, the ANI DN will be built with the ADDG. If RDNL ¦ 0, its value will be the number of digits extracted from the CLID to be used as the ANI DN. The least significant digits of the CLID will be extracted (for example, if the DN is 4201, the 1 is the least significant digit.)

If there is no CLID, the ANI DN of the Q931 route is used to build the ANI message. If there is no ANI DN on the Q931 route, the ANI DN of the CIS outgoing route is used to build the ANI message. If there is no ANI DN on the CIS outgoing route, the ANI is built with the ADDG of the CIS outgoing route (ADDG is always defined).

QSIG Gateway

The ANI information transmitted for this incoming QSIG route will include the LEC of the CIS outgoing route, the ANI DN, and the CAC of this incoming route.

The ANI DN information which is built will refer to the Calling Line Identification (CLID) if present and the Route DN Length prompt for ANI (RDNL ¦ 0) in LD 16. If the CLID is available but RDNL = 0 for that route, the ANI DN is the ANI DN of that incoming route. If the CLID is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, the ANI DN is the ANI DN of the CIS outgoing route. If the CLID is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, and the ANI DN of the CIS outgoing route is not defined, the ANI DN will be built with the ADDG. If RDNL ¦ 0, its value will be the number of digits extracted from the CLID to be used as the ANI DN. The least significant digits of the CLID will be extracted (e.g., if the DN is 4201, the 1 is the least significant digit.)

If there is no CLID, the ANI DN of the QSIG route is used to build the ANI message. If there is no ANI DN on the QSIG route, the ANI DN of the CIS outgoing route is used to build the ANI message. If there is no ANI DN on the CIS outgoing route, the ANI is built with the ADDG digits of the CIS outgoing route (ADDG is always defined).

The ANI information transmitted for this incoming QSIG route will include the LEC of the CIS outgoing route, the ANI DN, and the CAC of this incoming route.

R2MFC Calling Number Identification

The incoming R2MFC CNI will not be tandemed if the call is outgoing to a CIS trunk. The ANI built will be the LEC of the outgoing CIS route, the ANI DN of this R2MFC incoming route if defined (otherwise it will be the ANI DN of the outgoing CIS route, or the ADDG digit), and the CAC of this incoming R2MFC route.

The category (CAC) used to build the R2MFC Calling Number Identification (CNI) for the analog, digital and Basic Rate Interface (BRI) sets is used to build the CIS ANI. The meaning of CAC is different between the R2MFC CNI signaling and the CIS signaling (analog BRI, and digital). R2MFC CAC prompt values are in the range of 0 to 10, and the default is 0. CIS CAC prompt values are in the range of 0 to 9, and the default value is 3.

If the MFC package is equipped, but not the CIST package, the CAC prompt uses the R2MFC range and default. If the CIST package is equipped (MFC package equipped or not) the CAC prompt uses the CIS range and default.

Speed Call

Speed Call on an E3W trunk will fail for toll calls. E3W trunks do not wait for the ANI request from the Public Exchange, that is expected to appear after the toll access code is dialed. The Public Exchange will not accept the call due to the failure to receive ANI information.

Virtual Network Services

Virtual Network Services is not supported on CIS trunks.

Feature packaging

The Three-Wire Analog Trunk – CIS feature is contained in Commonwealth of Independent States Trunk Interface (CIST) package 221.

The following packages are also required to implement this feature:

- Fast Tone and Digit Switch (FTDS) package 87 (only for E3W cards)
- Flexible Tones and Cadences (FTC) package 125
- International Supplementary Features (SUPP) package 131 for DID/DOD
- Flexible Numbering Plan (FNP) package 160
- Trunk Failure Monitor (TFM) package 182, and
- Meridian 1 Extended Peripheral Equipment (XPE) package 203 (only for X3W cards).

Feature implementation

This is an example that describes how the 3WT related features are configured. Only the prompts that are significant for the Three-Wire Analog Trunk – CIS feature are mentioned.

The following features are needed to make the feature work according to this example: B34 Codec Static Loss Plan Downloading; Partial Dial Timer; End-of-Selection Busy; Tone to-Last Party; Special Dial Tones After Dialed Numbers; Trunk Barring, and Special Service List.

LD 17 – Configure the system data.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	PARM	Release 19 gate opener.
- PCML	A	System Pulse Code Modulation companding law. A-law is to be used in the CIS market.
...		
- DTRB	70	Dual-tone Multifrequency burst and interdigit pause for the Tone and Digit Switch. Pulse/Pause Ratio 70/70. For outgoing E3W cards, the preferable digitone burst time is 70 ms.

LD 16 – Configure an incoming X3W DID route.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	DID	Direct Inward Dialing trunk data block.
...		
DTRK	NO	This is not a digital trunk route.
...		
ICOG	ICT	Incoming trunk.
...		
CNTL	YES	Change control or timers.
- TIMR	ICF 0	Incoming flash timer should be set to 0. Validation is performed by 3WT firmware.
- TIMR	GTI 128	Incoming guard timer.
- TIMR	EOD 13952	End of dial timer, default value in milliseconds.
- TIMR	DSI 11904	Disconnect supervision timer in milliseconds.
- TIMR	DDL 0	Delay Dial Timer not needed.
...		
NEDC	ORG	Near End Disconnect Control. Originating end control.
FEDC	ORG	Far End Disconnect Control. Originating end control.
CDPC	(NO)	Meridian 1 is not the controlling party on incoming calls.
...		
OPR	(NO)	This is not an outpulsing route.

PRDL	YES	Partial dial timing is equipped using EOD.
EOS	BSY	Busy signal is sent on time-out.
DNSZ	(0)-7	Number of digits expected on DID routes. 0, the default, indicates no fixed value. This value must be defined according to the numbering plan.
...		
BTT	30	Busy Tone Time. Length of Busy/overflow to be returned on DID routes in seconds.
...		
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

LD 16 – Configure an outgoing X3W DID route and define the toll digit using the TDG prompt.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	DID	Direct Inward Dialing trunk data block.
...		
DTRK	NO	This is not a digital trunk route.
...		
ICOG	OGT	Outgoing trunk.
...		

CNTL	YES	Change control or timers.
- TIMR	ATO 128-(4992)-65408	ANI time out timer in milliseconds. For CIS outgoing trunk routes this defines the time delay performed after the outpulsing of the toll access code.
- TIMR	OGF 0	Outgoing flash timer should be set to 0 in milliseconds. Validation will be done by 3WT firmware.
- TIMR	EOD 13952	End of dial timer, default value.
- TIMR	DSI 11904	Disconnect supervision timer.
- TIMR	DDL 0	Delay Dial Timer not needed.
- TIMR	GTO 2944	Outgoing guard timer.
...		
NEDC	ETH	Near End Disconnect Control Either end control.
FEDC	ETH	Far End Disconnect Control Either end control.
...		
NATL	NO	North American Toll scheme.
TDG	8	Toll Digits. List of digits after trunk access code which indicate toll calls.
...		
OPR	(NO)	This is not an outpulsing route.
...		
ACKW	(NO)	Seizure acknowledge signal is not expected.
...		
LEC	0-9999999	Local Exchange Code. A value must be entered.
ADDG	0-(8)-9	Additional digit.
CAC	0-(3)-9	Route ANI category.

ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

LD 18 – Configure the Special Service List.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	SSL	Special Service List data block.
CUST	0-99	Customer number.
SSL	1-15	List number for Special Service List.
SSDG	xxxx	Special Service Digit or Digits (1 to 4 digits).
...		
- TOLL	YES	The SSDG entry is a toll number.
...		
SSDG	xxxx	Special Service Digit or Digits (1 to 4 digits).
...		
- SSUC	YES	The SSDG entry is a Special Service unanswered call.
SSDG	<CR>	

LD 16 – Configure an outgoing X3W DID route and define the toll access code using the SSL prompt.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	DID	Direct Inward Dialing trunk data block.
...		
DTRK	NO	This is not a digital trunk route.
...		
ICOG	OGT	Outgoing trunk.
...		
CNTL	YES	Change control or timers.
NEDC	ETH	Near End Disconnect Control Either end control.
FEDC	ETH	Far End Disconnect Control Either end control.
...		
SSL	1	Special Service List number.
...		
LEC	0-9999999	Local Exchange Code.
ADDG	0-(8)-9	Additional digit.
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

LD 16 – Configure an incoming E3W DID route.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	DID	Direct Inward Dialing trunk data block.
...		
DTRK	NO	This is not a digital trunk route.
...		
ICOG	ICT	Incoming trunk.
...		
CNTL	YES	Change control or timers.
- TIMR	ICF 0	Incoming flash timer should be set to 0. Validation has already been done by 3WT firmware.
- TIMR	OGF 0	Outgoing flash timer should be set to 0. Validation has already been done by 3WT firmware.
- TIMR	EOD 13952	End of dial timer, default value.
- TIMR	DSI 11904	Disconnect supervision timer.
- TIMR	DDL 0	Delay Dial Timer not needed.
...		
NEDC	ORG	Near End Disconnect Control Originating end control.
FEDC	ORG	Far End Disconnect Control Originating end control.
CDPC	(NO)	Meridian 1 is not the controlling party on incoming calls.
...		
OPR	(NO)	This is not an outpulsing route.

PRDL	YES	Partial dial timing is equipped using EOD.
EOS	BSY	End of selection and busy signals enabled.
DNSZ	(0)-7	Number of digits expected on DID routes. 0, the default, indicates no fixed value. This value must be defined according to the numbering plan.
...		
BTT	30	Length of busy/overflow tone to be returned on DID routes in seconds.
...		
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

LD 16 – Configure an outgoing E3W COT route.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	COT	Central Office Trunk data block.
...		
DTRK	NO	This is not a digital trunk route.
...		
ICOG	OGT	Outgoing trunk.
...		
CNTL	YES	Change control or timers.

- TIMR	ICF 0	Incoming flash timer should be set to 0 in milliseconds. Validation will be done by 3WT firmware.
- TIMR	OGF 0	Outgoing flash timer should be set to 0 in milliseconds. Validation will be done by 3WT firmware.
- TIMR	EOD 13952	End of dial timer, default value.
- TIMR	DSI 11904	Disconnect supervision timer.
- TIMR	DDL 0	Delay Dial Timer not needed.
- TIMR	GTO 2944	Outgoing guard timer.
...		
NEDC	ETH	Near End Disconnect Control Either end control.
FEDC	ETH	Far End Disconnect Control Either end control.
CDPC	(NO)	Meridian 1 is not the controlling party on incoming calls.
...		
NATL	NO	North American Toll scheme.
...		
LEC	0-9999999	Local Exchange Code.
ADDG	0-(8)-9	Additional digit.
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

LD 14 – Add or change trunk data for X3W incoming DID trunk.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DID	Direct Inward Dialing trunk data block.
...		
XTRK	XDID	Extended Trunk Type. IPE DID trunk card.
...		
SIGL	CIS	Trunk Signaling. Three-wire CIS trunk signaling.
CIST	(NO) YES	Prompted only for incoming routes (i.e., ICOG = ICT). NO = Local trunk. YES = Toll trunk.
...		
STRI	IMM	Immediate incoming start arrangement.
...		
SUPN	YES	Answer and disconnect supervision required.
CLS	(DIP)	Dial pulse (for 3WT incoming and outgoing).
	(SHL) LOL	Line length used for pad settings.
	(BARD) BARA	Barring (denied) allowed.

LD 14 – Add or change trunk data for X3W outgoing DID trunk.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DID	Direct Inward Dialing trunk data block.
...		
XTRK	XDID	IPE DID trunk card.
...		
SIGL	CIS	Three-wire CIS trunk signaling.
...		
STRO	IMM	Immediate outgoing start arrangement.
...		
SUPN	YES	Answer and disconnect supervision required.
CLS	(DIP)	Dial pulse (for 3WT incoming and outgoing).
	(SHL) LOL	Line length used for pad settings.
	(BARA) BARD	Barring (allowed) denied.

LD 14 – Add or change trunk data for E3W incoming three-wire trunk.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DID	Direct Inward Dialing trunk data block.
...		
SIGL	EAM	Ear &mouth.
CDEN	DD	Double density.
...		
STRI	IMM	Immediate incoming start arrangement.
...		
SUPN	YES	Answer and disconnect supervision required.
CLS	(DIP)	Dial pulse.

LD 14 – Add or change trunk data for E3W outgoing three-wire trunk.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	COT	Central Office Trunk data block.
...		
SIGL	LOP	Loop start.
CDEN	DD	Double density.
...		
SUPN	YES	Answer and disconnect supervision required.
- STYP	PSP	Polarity sensitive card.
...		

SEIZ	YES	Answer and disconnect supervision required.
CLS	DTN	Digitone.

LD 10 – Add or change analog (500/2500 type) telephones for CIS.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	500	analog (500/2500 type) telephone data block.
...		
CLS	(DNAA) DNAD	DN of set (allowed) denied for use in ANI messages.
CAC	0-9	Specifies ANI category for 3WT calls.

LD 11 – Add or change Meridian 1 proprietary telephones for CIS.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
...		
CLS	(DNAA) DNAD	DN of set (allowed) denied for use in ANI messages.
CAC	0-9	Specifies ANI category for 3WT calls.

LD 12 – Add or change an Attendant Console for CIS.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	ATT 1250 2250	Console type.
...		
CLS	(DNAA) DNAD	DN of set (allowed) denied for use in ANI messages.
CAC	0-9	Specifies ANI category for 3WT calls.

LD 27 – Add or change BRI sets for CIS.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DSL	Digital Subscriber Loop data block.
...		
CLS	(DNAA) DNAD	DN of set (allowed) denied for use in ANI messages.
CAC	0-9	Specifies ANI category for 3WT calls.

LD 56 – Configure dial tone, busy tone, and tone to last party.

Prompt	Response	Description
REQ	NEW CHG PRT	Add, change, or print.
TYPE	MCAD	Master Cadence data block.
WACD	30	Cadence number. In this example entry 30 is modified.
CDNC	60 60	On-off phases for cadence.
REQ	NEW CHG PRT	Add, change, or print.
TYPE	FCAD	Firmware Cadence data block.
WACD	30	Cadence number. In this example entry 30 is modified.
CDNC	60 60	On-off phases for cadence. 0.3 second on, 0.3 second off.
END	REPT	Repeating cycles.
- CYCS	1	On/off cycles to be repeated.
- WTON	YES	Define tones associated with the cadence.
- - TONES	158	420 Hz and -12 dB below overload.
REQ	NEW CHG PRT	Add, change, or print.
TYPE	FTC	Flexible Tones and Cadences data block. Used to provide special dial tone after dialed number.
...		
HCCT	YES	Hardware Controlled Cadences and Tones modification of the hardware.
...		
- BUSY		Busy tone.

- - TDSH		
- - XTON	158	420 Hz and -12 dB below overload.
- - XCAD	30	XCT cadence number. 0.3 seconds on, 0.3 seconds off.
...		
- TLP		Tone to last party.
-- TDSH		
- - XTON	158	420 Hz and -12 dB below overload.
- - XCAD	30	XCT cadence number. 0.3 seconds on, 0.3 seconds off.
- TLTP	30	Tone to last party timer in seconds.
...		
SRC	YES	Source Tones.
- SRC1		CIS continuous dial tone within the range.
- - TDSH		
- - XTON	158	420 Hz and -12 dB below overload.
- - XCAD	0	No cadence.
REQ	NEW CHG PRT	Add, change, or print.
TYPE	DTAD	Special Dial Tone After Dialed Number data block.
DDGT	9	The digit 9 is to be used as an outgoing local access code.
TONE	SRC1	Tone to be provided after the dialed digit 9.

LD 88 – Configure the Authcode data block.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	AUB	Authcode data block.
...		
CLAS	(0)-115	Classcode value assigned to Authcode (NAUT).
...		
NCOS	(0)-99	Network Class of Service Group number.
CAC	0-9	Specifies ANI category for CIS calls.

LD 97 – Configure the IPE system record for three-wire trunks.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	LOSP	Loss Plan Tables. Configure loss parameters for downloading.
...		
TTYP	(STAT)	Install a B34 Static Loss Plan Table.
- STYP	(PRED)	A numbered predefined table is to be used.
- - TNUM	28	28 = CIS Table.
REQ	CHG	Change.
TYPE	LOSP	Loss Plan Tables. Configure loss parameters for downloading.
...		
TTYP	(STAT)	Install a B34 Static Loss Plan Table.
- STYP	CSTM	Customize a numbered predefined table.

PWD2	xxxx	Response CSTM at STYP prompt requires a PWD2 password or a LAPW password with Loss Planning Customizing Allowed (LOSA) access. This prompt appears if the appropriate password has not been given previously.
- DIDS	Rx Tx	Enter loss levels for DID short line.
- DIDL	Rx Tx	Enter loss levels for DID long line.

Feature operation

No specific operating instructions are required to use this feature.

Introduced in X11 Release:	All
Networking:	No

Time and Date

The Time and Date feature provides the capability to display or modify the system time and date from the Attendant Console. If Display Time or Display Date keys are installed on the console, pressing the respective key causes the time or date to be shown on the digit display. However, these keys only allow information to be displayed, not changed.

The Change Time or Change Date keys allow the attendant to change the time or date. When a change is made, the system clock is altered to the new values. The change keys also allow display of the time or date.

Operating parameters

The Time and Date feature is available with QCW, M1250, and M2250 consoles.

If the Change Time (MTM) and Change Date (MDT) keys are provided on a console, there is no need to for the Display Time (DTM) and Display Date (DDT) keys because the MTM and MDT keys provide the display capability. DTM and DDT keys are used when the console is only allowed to view, but not change, the time and date.

When using the MTM and MDT keys, the date must be entered in the day, month, and year format; and the time must be entered in the 24-hour clock format. This is true even if the M1250 or M2250 has selected a different date and time format.

The M1250 and M2250 consoles continuously show the time and date on line one of the display. The attendant can change the format of time and date by using the Options menu.

The M1250 attendant can also change the date and time by using the Options menu. However, this only changes the time and date on the console and does not change the system clock. The MTM and MDT keys are required to change the system clock.

The date and time are downloaded to the M2250 console from the system clock and cannot be changed by the Options menu. The change time and date keys are required.

A call cannot be answered while the display/change key is activated; however, the keys can be used once the call is established.

Feature interactions

Hold

Loops used when updating time or date cannot be put on hold.

In-Band Automatic Number Identification

If the agent presses the Time and Date (TAD) key while on an In-Band Automatic Number Identification (IANI) call, the time and date remain displayed throughout the call. To display the ANI number again, place the call on hold and retrieve it. The ANI number reappears.

Network Time Synchronization

As done with the LD 12, every time the Time and Date Attendant key is used to change the system time, a request for synchronization will be made to the Master to accurately set the seconds.

Feature packaging

Time and Date (TAD) package 8 has no feature package dependencies.

Feature implementation

LD12 – Assign Time and Date keys on Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT 1250 2250	Console type.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx DDT xx DTM xx MDT xx MTM	Add a Display Date key. Add a Display Time key. Add a Display/Change Date key. Add a Display/Change Time key. ***Note:*** The range of key numbers (xx) is 0-19 on the M2250 console, and 0-9 on all other consoles.

Feature operation

To view the Time, press **Display Time (DTM)**.

To view the Date, press **Display Date (DDT)**.

To change the time, follow these steps:

1. Select an idle loop key.
2. Press **Change Time (MTM)**.
3. Enter the time using the 24-hour clock for hours and minutes (00 00).
4. Press **Change Time (MTM)**.
5. Press **Rls**.

To change the date, follow these steps:

1 Select an idle loop key.

2 Press **Change Date (MDT)**.

3 Enter the date using two digits for day, month, and year (dd mm yy).

4 Press **Change Date**.

5 Press **Rls**.

Tone to Last Party

This feature allows a Tone to Last Party (TLP) tone to be sent to analog (500/2500 type) telephones or trunks that are in the half disconnect state. The TLP is given until the Meridian 1 system releases the trunk, or the TLP timer (0-32 seconds) times out.

During the time that the TLP tone is given to the telephone, the telephone appears busy to all incoming calls. Camp-on is denied, and attendant Break-in, busy verify, and override are temporarily denied during this time.

If a telephone is not placed on-hook and the timer times out, the telephone is set in line lockout state, and remains so until it is placed on-hook.

A trunk is in the half disconnect state if the near-end has disconnected, but the Meridian 1 is still holding the trunk, waiting for a message from the far-end, or for the disconnect supervision timer to time out. Barge-in is denied while the trunk is receiving the TLP tone.

The TLP is defined in each tone table. The TLP for analog (500/2500 type) telephones is defined on a customer basis, while the TLP for trunks is defined on a route basis.

Operating parameters

The TLP tone is not given to a telephone that is receiving another tone.

This feature does not apply to service trunks, such as music, paging and recorded announcement.

The TLP tone is not given to a trunk if it is being held because of the guard timer.

Feature interactions

Multi-Party Operations

The TLP tone is not given to a telephone which has Multi-Party Operations (MPO).

Operator Call Back China #1

Operator Call Back China #1 (OPCB) has precedence over TLP.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 56 – Modify or change customer's tone and ringing parameters:

Prompt	Response	Description
...		
TLP	ccc ttt x xx xx xx	Tone to Last Party.
TLPT	(0)-32	Tone to Last Party Timer (seconds). No tone is given if TLPT = 0.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	All
Networking:	No

Tones and Cadences

A tone is the frequency and level of the sound produced while the telephone is ringing, providing dial tone, or providing feature activation tones. A cadence defines the time duration for the on and off phases of a ringing or tone cycle.

A set of basic tones and cadences is available on all systems. Flexible Tones and Cadences (FTC) package 125 in X11 Release 16 allows the tones to be changed.

Basic Tones and Cadences

Special dial tone

Special dial tone is supplied by the system to indicate a request for Call Transfer, Conference, and Ring Again. Special dial tone differs from regular dial tone in that it has three 128 ms interruptions at the beginning of the tone.

Overflow tone

Overflow tone can be provided on an optional basis to a station user who tries to access a trunk group when all trunks are busy, or who attempts to access features that are unavailable to their telephone. Overflow tone is best described as a fast busy signal.

Tone buzzing

Tone buzzing is used in conjunction with such features as Call Waiting and Manual Signaling (Buzz) to alert the user by a buzz tone through the telephone's loudspeaker. This applies when the telephone is off-hook or has a headset plugged in.

Flexible Tones and Cadences

X11 Release 16 introduces the Flexible Tones and Cadences (FTC) feature, allowing the system to adapt to tone specifications of different countries. Tones such as dial, special dial, busy, ringback, overflow, test, normal, and distinctive ringing are hardware controlled from the Tone and Digit Switch (TDS) circuit card (see Table 148). Tones such as camp-on, call waiting, intrusion, and override are software controlled, although the basic tone is still coming from the TDS card (see Table 149).

The desired cadences for the software controlled tones are defined by providing the system with the time length of the on and off phases. Software also controls ringing for analog (500/2500 type) telephones, although the voltage is supplied by the ring generator card.

The tone data is stored in tables. Every customer and route must select which tone table to use. Table 0 is filled in with default hexadecimal codes when the first customer is created and must not be changed.

All data related to the flexible tones is kept in isolated areas called Flexible Tone tables. Software Cadence tones and Master Cadence tables have an index into the MCAD table for its corresponding software cadence.

Most of the cadences are expressed in multiples of five milliseconds (ms). Therefore, in addition to the existing 128 ms timing mark, a 96 ms timing mark is introduced by a new read only memory (ROM) pack with new firmware.

Refer to *Flexible Tone and Digit Switch cards description* (553-2711-180) for complete details.

Operating parameters

The tones that can be produced are limited to the tones available on the particular TDS card being used.

Gradual level change is not allowed when a tone is activated.

If the Distinctive Ringing package is equipped, and a trunk route is classmarked for that feature, the cadence chosen for each call comes from the same tone table as for a normal call. The Distinctive Ringing field determines the cadences.

If a parked call was originally distinctive, and FTC is equipped, the Call Park Recall cadence takes precedence. If FTC is not equipped, the distinctive precedence ringing is given.

Because Enhanced Flexible Tones and Cadences (EFTC) is an enhancement to Flexible Tones and Cadences (FTC), the FTC package must be equipped.

A customer option determines whether the cadence will be defined by the originating or the terminating end of the call.

Feature interactions

Audible Reminder of Held Call

This feature allows for a definable cadence as a reminder of a held call. With an analog (500/2500 type) telephone, the cadence is determined by the customer's Flexible Tones and Cadence (FTC) table for the holding party. Ringing on an analog (500/2500 type) telephone is not affected by definitions for the Incoming Route option. The cadence for the reminder, and the duration between reminder rings, is always defined within the customer's tone table.

Call Forward Reminder Tone

The Call Forward Reminder Tone feature provides a way to determine whether the call forwarding feature on an analog (500/2500 type) telephone is active. For systems equipped with the FTC package, the Call Forward Reminder Tone Allowed option gives the dial tone defined by CFDT to a 500 or 2500 telephone that has CFW active with no message waiting and the dial tone defined by CFMW to a 500 or 2500 telephone that has CFW active and a message waiting. To get different Call Forward and Call Forward Message Waiting reminder dial tones, it is necessary to define a distinct tone and cadence for CFDT and a distinct tone and cadence for CFMW in LD 56, as well as to specify Call Forward Reminder Tone Allowed in LD 15.

Call Park Recall and Group Call Ring

Recall Ring and Group Call Ring are given special entries in the FTC table. New entries are added to the FTC overlay (LD 56) to define the cadence for Meridian 1 proprietary telephones, and analog (500/2500 type) telephones. The new Recall Ring entry is used to ring a telephone when recalling a Parked Call.

Conference Warning Tone Enhancement

There are no changes to the limitations to cadence numbers entry values. The same restriction still applies.

Ringing Based on Incoming Route

Enhanced Flexible Tones and Cadences (EFTC) allows the route's tone table to determine the cadence and ringing frequency for incoming calls.

10-Phase Cadence

Programming of software controlled cadences expands with EFTC from 4 intervals to 10, offering greater versatility with the cadences and cadence phases. This affects all cadences under software control.

Feature packaging

Flexible Tones and Cadences (FTC) package 125 has no feature package dependencies.

Feature implementation

Refer to *Flexible Tone and Digit Switch cards description* (553-2711-180).

Table 148
Hardware controlled tones (Part 1 of 2)

Tone	Description
Dial tone	Indicates the system can accept dialing.
Message Waiting dial tone	Indicates a message is waiting at the message center.
Call Forward dial tone	Indicates that the user has call forwarded the phone.
Call Forward Message Waiting dial tone	Indicates that the user has call forwarded the phone and a message is waiting at the message center.
Control Dial tone	Used for broker service to indicate a control digit is required after the switchhook (only for 2500-type telephones with Digitone class of service).
Busy tone	Indicates that the called DN is busy.
Ringback tone	Given to the calling party while the called party is ringing. Also given to Central Office trunks waiting for the DN to answer.

Table 148
Hardware controlled tones (Part 2 of 2)

Tone	Description
ACD RGA Ringback tone	Given to a caller to an Automatic Call Distribution (ACD) group when entering the waiting call queue and having RGA (Ring Again).
Overflow tone	Indicates that the trunk route is busy, or the DN is blocked or disabled, or that a not-allowed action has been carried out.
LDN tone	Indicates to a Centralized Attendant Service (CAS) attendant that the incoming call is a Listed DN (LDN) call from a remote site.
Camp-On tone	Provided as an initial burst when the attendant extends a call to a busy DN that is not equipped with the Call Waiting feature.
Camp-On Confirm tone	Confirms to a CAS attendant that a call to a busy DN at remote site has camped on, or that the called DN has not answered after a specified time and the calling party has come back.
Dial "0" Recall tone	Indicates to a CAS attendant that a call is a recall occurring due to attendant recall or call forward busy to an attendant from a remote site.
Hold Confirm tone	Indicates to a CAS attendant that a call placed on silent hold has timed out and is recalling.
Test tone	Provided during testing of trunk circuits.
Distinctive Ring tone	Used to differentiate between routes.
Normal Ring tone	Provided for internal calls and incoming calls if distinctive ringing or precedence ringing is not in use.

Table 149
Software controlled tones

Tone	Description
Agent Observe tone	Given to an agent being observed by a supervisor.
Call Waiting tone	Indicates to a busy station that another call is coming in.
Intrusion tone	Provided when the attendant initiates the Barge-In, Busy Verify, or Break-In feature.
Override tone	Provided when a user operates the Override key and enters the conversation of a busy extension.
Observe Blocking tone	Given to the supervisor who encounters blocking while attempting to observe an agent.
Off-Hook Queuing tone	Given to the call originator when the call enters the off-hook queue.
Set Relocate tone	Given after all information needed to relocate the phone is given and proven to be correct. Also given to indicate all is correct after plugging the phone back in at the relocated Terminal Number (TN).
Telset Messaging Alert tone	Indicates to caller that Telset messaging facilities have been entered.
Telset Messaging OK tone	Indicates to caller that the message has been received correctly and everything is fine.
Tel Status Update tone	Indicates a successful status update process.
Special Dial tone	Indicates the availability of a special function such as Conference, Transfer, etc.
Expensive Route Warning tone	When Automatic Route Selection is in use, indicates that all inexpensive routes are busy and an expensive route must be chosen to complete the call.
ACD Call Force tone	Indicates to the ACD agent that the current call has been disconnected and a new caller is about to be given to the agent.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	14
Networking:	No

Tones, Flexible Incoming

When a telephone is off-hook, the user is alerted to a second incoming call by a buzz tone. Flexible Incoming Tones (FIT) allows the replacement of the standard buzz tone with a buzz with an on/off cadence. This feature is defined on an individual telephone basis.

When a call is presented to a telephone in any of the following situations, a tone with a special cadence alerts the user:

— Call on DN key while busy on another DN

— Call to a station that is off-hook

— Call Park recall when station is busy on another DN

— Call on Group Call key while busy on another call

— Call Waiting, and

— Call on Dial Intercom key while busy on another call.

The buzz cadence is the same as the ringing cadence that applies to a particular kind of call. For example, if a user receives a call that is a Group Call, FIT alerts users with a buzz cadence unique to group calls. If the user receives a call on the Call Waiting key, FIT provides a buzz cadence signifying call waiting.

Operating parameters

Flexible Incoming Tones applies only to Meridian 1 proprietary telephones.

Flexible Incoming Tones does not apply to the following:

— Automatic Call Distribution (ACD) call forcing

— ACD agent receiving a call on ASP key

— ACD supervisor receiving a call on AMG key

— Manual signaling

— Signal Source activated by an Attendant Console, and

— Ring Again.

Digital telephones in Handsfree mode receive the regular buzz, even if FIT is enabled.

The telephone buzzes with a cadence only if the customer and telephone options are activated. If either option is off, the telephone receives the standard buzz.

Feature interactions

Automatic Call Distribution

If an Automatic Call Distribution (ACD) agent telephone has FIT allowed and either is off-hook in the handset mode or has the headset plugged in, the agent receives a buzz cadence when a new call is presented. If FIT is not allowed, the agent telephone receives the standard buzz tone.

Dial Intercom Groups

For Dial Intercom Group (DIG) calls with the voice (V) option, if the telephone receiving the call is busy, the user hears one buzz followed by a flashing indicator. This is how DIG works with or without FIT.

Feature packaging

Flexible Incoming Tones is included in base X11 system software.

Feature implementation

LD 15 – Allow or deny Flexible Incoming Tones (FIT) at the customer level.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
CUST	0-99	Customer number.
- OPT	(SBD) SBA	FIT (denied) allowed for SL-1 sets.
	(DBD) DBA	FIT (denied) allowed for Meridian digital telephones.

LD 11 – Allow or deny Flexible Incoming Tones for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	(FITD) FITA	Flexible Incoming Tone (denied) allowed.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	15.59F
Networking:	Yes

Total Redirection Count

This feature allows a limit to be defined on the number of redirections within a private network allowed to a call, before the call receives intercept treatment.

Both the limit on the redirection for a call and the type of Intercept treatment that the call receives are customer-defined in LD 15. This applies to on-node and off-node redirections, and to all types of redirections.

Operating parameters

The maximum value that may be given to the Total Redirection Count (TRCNT) limit is seven.

The TRCNT is kept active until the call is established or directed to the attendant.

The TRCNT takes precedence over higher count limits placed on redirected calls, while lower count limits take precedence over the TRCNT.

It is possible to define a different TRCNT limit at each node. For this reason, it is possible for a node to receive a redirected call from another node that exceeds its TRCNT limit. In this case, the TRCNT count for the call is set to the TRCNT limit defined for the node. At least one attempt is made to terminate the call before intercept treatment is given.

For off-node operation, the TRCNT count overrides the Redirection Count (RCNT) count in the Integrated Services Digital Network (ISDN) field in the SETUP message. This implies that the count transmitted to a node is either interpreted as TRCNT or Call Redirection Threshold (RCNT), depending on the configuration at the receiving node.

For off-node calls, this feature applies only to Meridian 1 systems using Meridian Customer Defined Networking (MCDN) signaling over ISDN Signaling Link (ISL)/ISDN TIE links. Network Attendant Service is required to route a call to an attendant at another node.

Intercept to the attendant does not count as a redirection attempt.

The following ISDN call restrictions apply:

- Tandem Threshold, which is the limit placed on the number of tandem nodes allowed in a network connection.
- The Public Service Telephone Network (PSTN) Threshold, which is the limit placed on the number of PSTNs allowed in a network connection.
- The Call Redirection Threshold, which is a limit on the number of times that a call can be redirected off-node. If the Total Redirection Count (TRCNT) Limit is set a value greater than zero, the ISDN field in the SETUP message transports the TRCNT information rather than the Redirection Count (RCNT) information.
- The Mμ/A Law Conversion Threshold, which is a limit on the number of Mμ/A Law Conversions allowed in a network connection.
- Satellite Delay Threshold, which is a limit on the number of satellite delays allowed in a network connection.
- Disconnect Supervision Threshold, which limits to one the number of unsupervised trunks allowed in a network connection.

Feature interactions

Call Forward No Answer and Transfer

If a call has attempted Call Forward No Answer and was extended by the attendant, the call will not be intercepted when the TRCNT limit has been exceeded. The call will continue to ring the set until recalled to the attendant.

Group Hunt

Group Hunt takes precedence over the TRCNT feature, in that the TRCNT limit is not applied to a Group Hunt call.

Hunt
Call Forward Busy
Call Forward All Calls
Call Forward No Answer
Second-level Call Forward No Answer

Hunt, Call Forward Busy, Call Forward All Calls, Call Forward No Answer, and Second-level Call Forward No Answer redirections are limited to the value defined in the TRCNT limit (if greater than 0). If this limit is exceeded, intercept treatment is given.

Intercept treatment

Intercept treatment is not given if a call is a Network Automatic Call Distribution (NACD) ACD call, if a call is a Central Office trunk in Night Service (specific treatment is given rather than customer-defined intercept treatment), or if the call is a data call (overflow tone is automatically given).

Feature packaging

For inter-node operation, Integrated Services Digital Network (ISDN) package 145.

For detecting trunk type across a network, Network Attendant Service (NAS) package 159.

For attendant display, Calling Party Name Display (CPND) package 95.

For the attendant to override a redirection configuration, Attendant Break-in/Trunk Offer (BKI) package 127.

Feature implementation

LD 15 – Configure the type of intercept treatment that the redirected call receives, and the number of times that a call can be redirected before being intercepted.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB INT	Customer Data Block. Release 21 gate opener.
...		
- RCLE	(ATN) OVF ATN	Redirection Count Limited Exceeded as defined by TRCL. ATN is not allowed for attendant calls. NAP is not allowed for any field for RCLE.
TYPE	RDR	Release 21 gate opener.
...		
- TRCL	(0)-7	Total Redirection Count Limit. Number of times that a call can be redirected before being intercepted. Zero means that redirection is not limited by this feature, but is limited by various configurations.

Feature operation

When the total redirection count exceeds the defined limit, the call receives the customer-defined intercept treatment. This treatment includes receiving busy indication, overflow indication, or recorded announcement, receiving one of eight special tones, or being routed to the attendant. If the call is routed to the attendant, it is presented on the Incoming Call Indicator (ICI) Intercept key and the reason for redirection is given on the console display. The attendant may then use Attendant Break-In to connect to the desired station (if the desired station is established on a call).

Introduced in X11 Release:	10.10B
Networking:	No

Trunk Barring

The Trunk Barring feature provides the option of denying or allowing a direct or modified connection between customer defined routes.

Trunk Barring works in conjunction with Route Access Restriction Tables (ARTs) defined in LD 56. Trunk Barring is applied on a route basis. The four route categories that Trunk Barring recognizes, and the types of routes in each category, appear in the following table:

Route Category	Route Types
Central Trunk Office (COT)	COT, FEX, WAT
Direct Inward Dialing	DID
TIE	TIE, CAA, CAM, CSA
Other trunk types	ADM, DIC, MDM, PAG, RCD

Trunk Barring applies to all methods of connecting the trunks (e.g., dialing route access, call modification, or attendant extension). A route is allocated an Access Restriction Table (ART) linked by a table number (ART number) in the Route Data Block. The ART to be used for a connection is determined by the first trunk in the connection independent of whether the trunks are incoming or outgoing. The first trunk in the connection is referred to as the Originating Trunk Connection (OTC).

A default table exists so that LD 56 does not have to be used to assign an ART number to a newly created route. If the default value for each Route Category is ART number 0, no trunk barring will occur.

Operating parameters

When activated in conjunction with the Route Access Restriction Tables, Trunk Barring prohibits previously allowed connections. Previously restricted connections cannot be lifted or circumvented by Trunk Barring.

Trunk Barring does not apply to Recorded Announcement (RAN), Music (MUS), Automatic Wake-Up (AWU), or Centralized Attendant Service (CAS) trunks as it is inconsistent with their defined purposes.

Feature interactions

Access Restrictions

Trunk Barring is at the top of the hierarchy for access restrictions.

Attendant Break-In

Trunk Barring does not result in intercept treatment for Toll Operator Break-In.

Attendant-Extended Calls

When an attendant attempts to extend an Originating Trunk Connection on a barred route, overflow tone is given.

Call Forward All Calls
Call Forward Busy
Call Forward by Call Type
Call Forward External Deny
Call Forward, Internal Calls
Call Forward No Answer
Call Forward No Answer, Second Level

If an Originating Trunk Connection is forwarded to a barred route, the caller receives the intercept treatment specified in the Customer Data Block.

Call Transfer

The originator of a call transfer, unless otherwise restricted, is able to connect to a denied party on a consultation basis. Operating the Transfer key on a Meridian 1 proprietary set or going on-hook on an analog (500/2500 type) set does not result in a call transfer if the Originating Trunk Connection is barred. The user of a Meridian 1 proprietary set remains connected to the denied party until releasing the connection and returning to the held Originating Trunk Connection. The user of an analog (500/2500 type) set is re-rung by the Originating Trunk Connection when a transfer is attempted and denied.

Conference

The originator of a conference call can only connect to a barred route on a consultation basis. A switchhook flash from an analog (500/2500 type) set results in a re-established connection with the Originating Trunk Connection. The user of a Meridian 1 proprietary set must release the barred connection to return to the Originating Trunk connection, or the conference containing the Originating Trunk connection; operating the Conference key on a Meridian 1 proprietary set has no effect. An attendant can return to the Originating Trunk Connection, or the conference containing the Originating Trunk Connection, by releasing the barred connection. This is done by pressing the RLS DEST key; pressing the Conference key has no effect.

Direct Trunk Access

When an Originating Trunk Connection attempts a trunk connection to a route which is restricted by its Access Restriction Table, the connection is not allowed. The intercept treatment specified in the Customer Data Block is applied.

Enhanced Night Service

Any incoming call that is routed by Enhanced Night Service to a set from which it is barred will not be connected. Overflow tone (fast busy) will be given to the incoming trunk instead.

Intercept Treatment

A telephone that is intercepted to the attendant cannot apply Ring Again on No Answer.

ISDN Semi Permanent for Australia

For calls using or requesting an ISPC link, Trunk Barring is provided according to the configuration of the route associated to the phantom trunk TN. This configuration is independent of the route associated to the real TN.

Network Alternate Route Selection (NARS)/Basic Alternate Route Selection (BARS)

If one route is barred, the system will look for the next route in the Route List Index (RLI) and if this route is not barred, the call will go through on this route. If the second route is barred, the system will continue searching the next route in the Route List Index, until an unbarred route is found.

When implementing Trunk Barring caution must be exercised not to circumvent the intended NARS/BARS restrictions.

Toll Operator Break-In

Trunk Barring results in intercept treatment for all route types that can be barred, except Toll Operator Break-In.

Virtual Network Services

With respect to this feature the following cases apply:

— When the second trunk involved in the call is used by VNS, no trunk barring is applied regardless of the configuration of the first trunk. The call is always allowed to get through.

  Note: This implementation completely overrides the Trunk Barring feature.

— When the first trunk involved in the call uses VNS, and the second one is not used by VNS, trunk barring is performed according to the content of the default ART table for the TIE trunk.

Feature packaging

Trunk Barring (TBAR) is package 132.

Feature implementation

In most cases that require barring, only one ART is necessary, although multiple ARTs may be defined per route. Whenever a new route is created (in LD 16), the default ART defined for that route type is assigned to the route. This default depends on the route type being created.

The flexibility of assigning ART by route is also available. The default table which specifies which ART table is to be assigned to a route type is changeable in LD 56. Until this is done, the default ART is used.

The following is a guideline on how to set up Trunk Barring

1 Gather all information regarding the type of route to be used in the Meridian 1 system.

2 For each route type, list beside it the route types that are barred from connecting to it.

3 For each route type, assign a code number from 1 to 63. Look for the route types that are barred from accessing the same types and assign the same code number to them. If a route type is not barred from accessing any other route type, it is assigned code number 0.

4 When each route type is assigned a code number, go back to step 2 and replace the route types that the route is barred from accessing with their code number.

5 Using LD 56, create all necessary Access Restriction Tables (ARTs). Using the code number of the originating route type as the ART number, deny the necessary route type using the code numbers assigned in the previous step.

6 Assign each ART to a route in one of two methods:

- **a** Use LD 56 to create the Route Category Default Table (RCDT). As each route is created using LD 16, it is assigned the default ART according to route type.
- **b** Use LD 56 to assign to existing routes the desired ART.

The following is an example of how to set up trunk barring using the procedures listed above. This example is not reflective of the typical situation, but is only used to show the steps involved.

List all route types.

- COT
- TIE
- DID
- PAG
- DIC
- RAN – ignore because it cannot be barred.
- MDM

List route types to which the originator is barred access.

- COT is not barred from accessing any type.
- TIE is barred from accessing COT, PAG, DIC, and DID.
- DID is barred from accessing TIE, DIC, and MDM.
- PAG cannot be originator, but can be barred by other route types.
- DIC cannot be originator, but can be barred by other route types.
- MDM is barred from accessing COT, DID, PAG, and DIC.

Assign each originating route type a code number from 0 to 63.

- COT is assigned 0 (it is not barred access to any route type).
- TIE is assigned 1.
- DID is assigned 2.
- PAG is assigned 0 (this cannot be an originating route, but it can be barred by other route types).
- DIC is assigned 0 (this cannot be an originating route, but it can be barred by other route types).
- MDM is assigned 3.

Replace the route types the originator is barred from accessing with their code numbers.

- COT (ART 0) – not barred.
- TIE (ART 1) – is barred from accessing 0, and 2.
- DID (ART 2) – is barred from accessing 0, 1, and 3.
- PAG/DIC (ART 3) – not barred.
- MDM (ART 4) – bars 0, and 2.

Set up the Route Category Default Table (RCDT).

- COT 0.
- TIE 1.
- DID 2.
- OTH 0 – MDM will initially be assigned ART 0 like DIC and PAG, but can be changed using the RART prompt.

LD 56 – Modify trunk barring Access Restriction Tables (ARTs).

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	TBAR	Add or change Access Restriction Table(s) (ARTs).
ART	(0)-63	Select ART to add or change. If ART table 0 is defined, no restrictions apply.
	<CR>	Return to REQ prompt.
DENY	yyy yyy	Enter ART numbers denied to Originating Trunk Connection (OTC).
	ALL	Deny all ARTs to OTC.
	xALL	All ART numbers allowed to OTC.
	Xyyy Xyyy	Enter ART numbers allowed to OTC, or change to remove previously blocked connections.
	<CR>	Return to REQ prompt with no table being stored.

LD 56 – Change, or print ART number for the route.

Prompt	Response	Description
REQ	CHG PRT	Change or Print. ***Note:*** REQ = NEW, or OUT is disallowed for RART.
TYPE	RART	Change ART number for the route.
CUST	(0)-99 (0)-31	Customer number. For Option 11 C.
ROUT	(0)-511 (0)-127	Route number. For Option 11C.
ART	(0)-63	ART to assign to route(s). If ART table 0 is defined, no restrictions apply.
	<CR>	Return to REQ prompt. ART remains unchanged.

LD 56 – Change, or print the route category default table.

Prompt	Response	Description
REQ	CHG PRT	Change or Print. ***Note:*** REQ = NEW, or OUT is disallowed for RCDT.
TYPE	RCDT	Change the route category default table.
COT	(0)-63	COT, FEX, and WAT routes are assigned the entered ART when the route is created in LD 16.
DID	(0)-63	DID routes are assigned the entered ART when the route is created in LD 16.
TIE	(0)-63	CAA, CAM, CSA, and TIE routes are assigned the entered ART when the route is created in LD 16.
OTH	(0)-63	ADM, DIC, MDM, PAG, and RCD routes are assigned the entered ART when the route is created in LD 16.
	<CR>	Return to the REQ prompt.

Feature operation

No specific operating instructions are required by the user. Barring is implemented via service change by a qualified technician. If the connection is not allowed, intercept treatment defined by the ACCD prompt in LD 15 is implemented.

Introduced in X11 Release:	15.58F
Networking:	No

Trunk Failure Monitor

The Trunk Failure Monitor (TFM) feature detects Line Break Alarm Signals (LBAS), which are generated because of trouble conditions on Direct Inward Dialing (DID), Direct Outward Dialing (DOD), or TIE trunks, or service degraded to Out-of-service (OOS) on 2 Mbps Digital Trunk Interface (DTI) or Primary Rate Interface (PRI) trunks. If a line break is detected, a trunk message is printed on the maintenance TTY, and the affected trunk is rendered BUSY to stop any further seizure of the trunk during outgoing calls.

Once the line break trouble condition has been fixed, a different Line Break Alarm Signal (LBAS) is generated. The TFM feature detects this signal, prints another trunk message on the TTY indicating that the trouble condition has been corrected, and renders the repaired trunk unit IDLE for normal use.

Operating parameters

TFM is not supported by the Attendant Administration feature.

TFM is not supported on 1.5 Mbps DTI.

A Centralized Attendant Service (CAS) attendant can only monitor the trunks on the switch on which the attendant is located.

This feature is supported on the following Attendant Console types only:

— QCW3

— QCW4

— M1250, and

— M2250.

Feature interactions

TFM requires the QPC730B for DID or DOD trunks, and the QPC774 for TIE trunks.

Extended DID/DOD Software Support - Europe

received from IPE XDID trunks are treated in the same manner as the EPE Line Break Alarm/Line Break Alarm Clear signals are treated for EPE trunks (LD 15 must be configured with TFDR = YES); when a BAR message indicating a problem situation is received, a TRK501 message is printed on the TTY, the uppermost key lamps light up on the Attendant Console, and the trunk is placed into BUSY state to prevent the trunk from being seized for new outgoing calls. The reception of an UNBAR message indicates that the problem situation has been cleared. A TRK502 message is printed on the TTY, the lamps on the Attendant Console are darkened, and the trunk is idled. Note that BARA CLS must be configured on the XDID trunk for the described process to occur.

Extended Flexible Central Office Trunk Software Support

As part of the Trunk Failure Monitor feature, the BAR/UNBAR messages received from IPE XFCOT trunks are treated in the same manner as the EPE Line Break Alarm/Line Break Alarm Clear signals are treated for EPE trunks. When a BAR message indicating a problem situation is received, a trunk message is printed on the TTY, the uppermost key lamps light up on the Attendant Console, and the trunk is placed into BUSY state to prevent the trunk from being seized for new outgoing calls. The reception of an UNBAR message indicates that the problem situation has been cleared. A message is printed on the TTY, the lamps on the Attendant Console are darkened, and the seized trunk is idled. Note that BARA Class of Service must be configured on the trunk for the described processing to occur.

Feature packaging

Trunk Failure Monitor (TFM) package 182.

Feature implementation

No change to existing configuration is required for the Trunk Failure Monitor feature.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	15.58F
Networking:	No

Trunk Failure Monitor Enhancement

This enhancement to the Trunk Failure Monitor feature provides a visual display on attendant consoles to indicate Direct Inward Dialing (DID)/Direct Outward Dialing (DOD)/TIE trunk line-break alarm conditions, and optionally to indicate 2 Mbps Digital Trunk Interface or Primary Rate Interface (PRI) Out-of-service conditions. The upper-most left key lamps on the console flash to indicate these trouble conditions.

This capability is available on the following Meridian Attendant Consoles:

— QCW3

— QCW4

— M1250, and

— M2250.

Operating parameters

Trunk Failure Monitor (TFM) package 182 must be equipped.

This enhancement is not supported for:

— Tenant Groups attendants

— 1.5 Mbps Digital Trunk Interface (DTI), and

— Automatic Trunk Maintenance.

Feature interactions

There are no interactions with other features.

Feature packaging

Trunk Failure Monitor (TFM) package 182.

Feature implementation

LD 15 – Configure the trunk failure display.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
...		
- TFDR	(NO) YES	Trunk Failure Display required. Prompted with TFM package 182. Requires QCW3, QCW, M1250, or M2250 consoles.

Feature operation

The upper-most left key lamps on the console flash to indicate trouble conditions.

If the attendant is in Position Busy, Night Service, or Loop Busy state, without a call on the console, pressing the upper-most left key causes the display to show the failed trunk unit or loop number. The lamp state changes from flashing to lit. If there is more than one failed trunk or loop, the display shows them one at a time, and the lamps remain flashing until all failed trunk units or loop numbers are displayed.

When the trouble conditions have been resolved, the lamps become dark to indicate that the trunk or loop is available for normal use.

Introduced in X11 Release:	22.0X
Networking:	Yes

Trunk to Trunk Connection

The Trunk to Trunk Connection feature introduces the following capabilities: transfer on ringing of external trunk across the network, transfer of one supervised outgoing external trunk to another, conference of external trunks and outgoing trunk to trunk charging. These capabilities are available on an analog (500/2500 type) set, Meridian 1 proprietary set or an Attendant Console.

Transfer on Ringing of External Trunk over Network

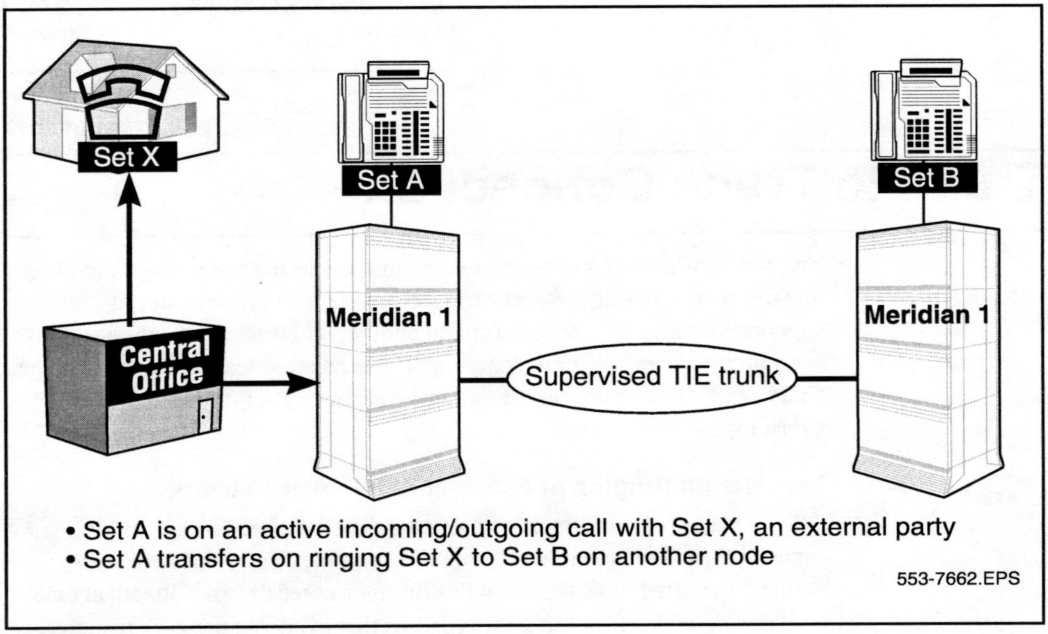
Allows the transfer on ringing of an established external trunk call over a supervised analog network TIE trunk. If the called party does not answer within a specified time, the call will slow answer recall to the attendant on the transferring node. This capability ensures that available network resources are not occupied indefinitely.

Transfer of External Trunks

Allows the transfer of one outgoing external trunk to another trunk provided both calls are answered and both trunks have answer supervision.

As illustrated in Figure 89, Set A is on an incoming/outgoing call with Set X, an external trunk. Set A initiates a call transfer of Set X to Set B. With the Trunk to Trunk Connection feature, Set A can transfer on ringing without waiting for Set B to answer. If Set B does not answer the transferred call, the external trunk slow answer recalls to the attendant on the transferring node.

Figure 89
Transfer on ringing of external call

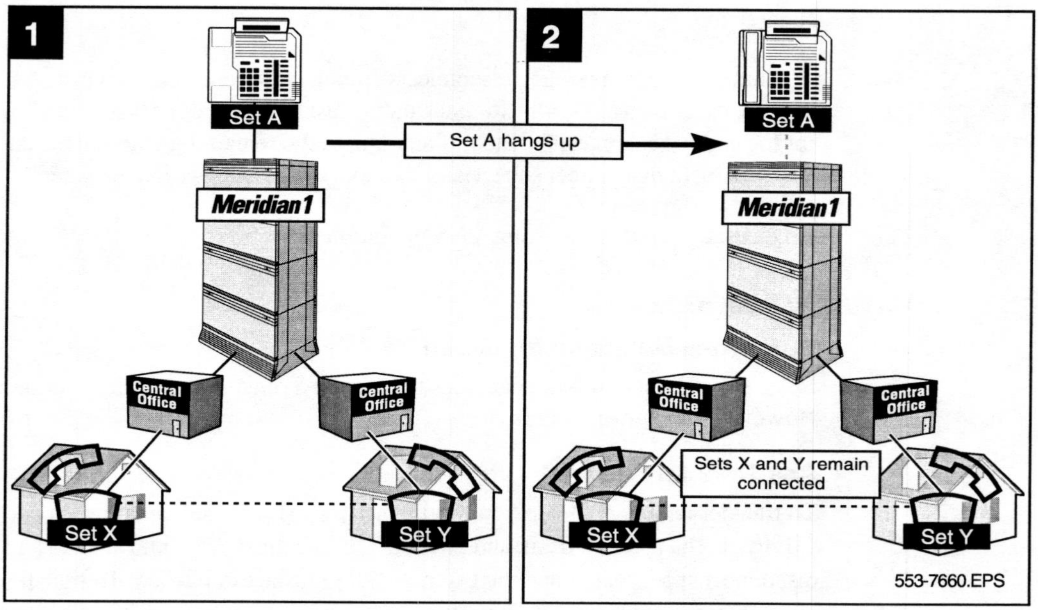


Conference of External Trunks

Allows external trunks to remain established in a conference call in circumstances when all external trunks involved in the call offer disconnect supervision.

Figure 90 illustrates the Conference of External Trunk capabilities of this feature. Set A is on an established conference call with two or more external trunks, Set X and Set Y. When Set A disconnects during the conference, Set X and Set Y continue in the established call.

Figure 90
Conference of external trunks



Outgoing Trunk to Trunk Charging

Ensures that outstanding charging information, relevant to both outgoing calls, is contained in relevant Call Detail Recording records.

Operating Parameters

The feature is applicable to Meridian 1 Options 11C, 51C, 61C, 81 and 81C systems.

Slow answer recall occurs when an external trunk is transferred on ringing across an answer supervised network TIE trunk to a set that does not answer. However, the resulting recall will be to an attendant on the transferring node and not to the original set which transferred the call.

When transferring one outgoing trunk to another, it is required that the two external calls involved are both answered prior to completing the transfer. Both external trunks involved must have both answer and disconnect supervision.

When the last internal party disconnects from a conference call, involving 2 or more external trunks, all external trunks must have disconnect supervision for the call to remain established. If any one of the remaining external trunks does not have disconnect supervision, all external trunks will be dropped.

No change is made to existing VNS operation.

Feature interactions

Busy Tone Detection for Japan

Busy Tone Detection for Japan does not impact Trunk to Trunk Connection. However, which ever occurs first prevails.

Call Transfer

To transfer an external trunk on ringing across a supervised analog network TIE trunk, the external trunk and internal TIE line must have both answer and disconnect supervision, and the external call must be established. To transfer one outgoing external trunk to another, both external trunks must have answer and disconnect supervision, and both external calls must be established.

Centralized Answering Position

The Option 11C system may not have an actual attendant console. Instead, the Option 11C will use Centralized Answering Position (CAP). The CAP Directory Number (DN) is the customer Night DN. Since no attendant is configured, the customer is viewed to be in Night Service and any calls for the attendant are directed to the CAP. Slow Answer Recall may be presented to a CAP when no attendant console is configured for the customer.

Conference

Trunk to Trunk Connection allows external trunks to remain established in a call, provided that all external trunks involved have disconnect supervision. With respect to charging costs associated with a conference call, once the last set involved in the conference call disconnects, a search is made of all remaining trunks in the call to determine which call is established in the call for the longest period of time. This trunk is the chargeable Terminal Number (TN). This process is repeated to find the next chargeable TN.

Multi-Party Operations - Ringing No Answer

In a standalone environment, the RGNA prompt in the Customer Data Block will be used when an external trunk is transferred on ringing and the called party does not answer. In a network environment, the RTIM timer value in the Customer Data Block will be used for slow answer recall.

Message Registration

The last party releasing the call collects the total value of outstanding Periodic Pulse Metering (PPM) generated on outgoing trunks. If the last party is an internal set, the outstanding PPM is stored against the meter of the set. If the last party is an internal TIE trunk, the outstanding PPM is stored against the meter associated with the internal TIE trunk access code. If the last party is an outgoing external trunk, the outstanding PPM is stored against the meter associated with the external trunk access code.

Night Service

If an attendant is placed in Night Service, calls to the attendant are directed to a station with the Night DN. Recalls are not directed to the Night DN. Recalls are put in the attendant call waiting queue when in Night Service.

Night Service Enhancement

Recalls made while the attendant is in Night Service are routed to the Night DN, if the original call is an external call. In such a case, the destination party is disconnected, the internal network trunk is released and the original extended call is presented to the Night DN. If the original call is internal, recalls are put in the attendant call waiting queue when in Night Service.

Trunk Barring
Trunk Group Access Restriction

Trunk Barring and Trunk Group Access Restriction takes precedence over the Trunk to Trunk Connection feature.

Feature packaging

Trunk to Trunk Connection is included in base X11 system software.

> *Note:* DID to TIE (DTOT) for Japan package 176 must be restricted to enable this feature.

Feature implementation

LD 15 – Modifications to Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	ATT	Attendant console prompts.
CUST	xx	Customer number.
RTIM	xxx yyy zzz	Enter defined value for the Slow Answer Recall timer where: xxx = 0-(30)-378 Slow Answer Recall yy y= 0-(30)-510 Camp On Recall zzz = 0-(30)-510 Call Waiting Recall

LD 15 – Modifications to Customer Data Block

Prompt	Response	Description
REQ:	CHG	Change
TYPE	NET	Trunk and network options.
CUST	xx	Customer number.
...		
ISDN	YES	Change the Integrated Services Digital Network options.
- PSTN	NO	Public Switched Telephone Network. Limit the number of PSTNs allowed in a network connection to one PSTN. NO= Puts no limit on the number of PSTN connections. YES = Limits the number of PSTN connections.
...		
DITI	YES	Allow Direct Inward Dialing to TIE connections for customer.
TRNX	YES	YES = Allows transfer on ringing of an external trunk over a supervised analog network TIE trunk across private network. NO= Prevents transfer on ringing of an external trunk over a supervised analog network TIE trunk across private network.
EXTT	YES	YES = Allows connection of supervised external trunks. NO = Prevents connection of supervised external trunks.

Feature operation

No specific operating procedures are required to use this feature.

Trunk Traffic Reporting Enhancement

The following modifications to trunk traffic reporting have been implemented to improve the accuracy of TFC002 traffic reports. The options are selected in the Configuration Data Block.

Traffic Period Option

Without enabling this option, trunk usage added its entire duration into the traffic period in which the disconnection occurred. If the duration was longer than 36 CCS (CCS = 100 call seconds), but less than 50 CCS, a TFS401 message was output. However, that duration was still accumulated and included in the traffic reports. If the duration was longer than or equal to 50 CCS, a TFS402 message was output. This duration was not accumulated, and was excluded from the traffic reports.

The Traffic Period Option enables the CCS to be reported in each traffic report interval. The peg count is still reported at disconnect time as per existing operation.

Note that when the Traffic Period Option is first enabled, the first traffic report may get some TFS403 messages.

Trunk Seizure Option

Without enabling this option, Meridian 1 traffic statistics began accumulating when a call was established. Meridian 1 software determined that the call was established when one of the following occurred: the End-of-Dialing (EOD) timer timed out after the last digit was dialed; the octothorpe (#) was dialed; or answer supervision was received. In some situations, customers could not match Meridian 1 traffic reports with their carrier reports, because many carriers start accumulating statistics when a trunk is seized.

The Trunk Seizure Option provides the ability to start accumulating statistics upon trunk seizure, rather than when the call is established.

Operating parameters

If the duration of a call is less than two to four seconds, the peg count is not accumulated. This functionality only applies when the trunk seizure option is enabled.

Due to the accumulation at trunk seizure, peg counts occur even if a call is unanswered.

Feature interactions

Automatic Call Distribution

A trunk call to an Automatic Call Distribution (ACD) DN will only be considered established once this call is answered. It is not considered established while the call is waiting in the ACD queue. Therefore, at the end of a traffic period, if a trunk call is in the ACD queue, the Traffic Period Option will not accumulate the duration for this call.

Note that when the duration is accumulated at disconnect or at the end of a traffic period after this call is answered, the total duration including the time the call was in the ACD queue is accumulated. This total duration may be longer than a single traffic period due to the time in the ACD queue and a TFS401, TFS402, or TFS403 message may be output.

Music

The Trunk Seizure Option is not supported on Music trunks.

Recorded Announcement

The Trunk Seizure Option is not supported on Recorded Announcement trunks.

Traffic Monitor

The Traffic Monitor feature outputs certain traffic data approximately every minute.

The trunk usage and peg count output by the Traffic Monitor feature can be enhanced by enabling the Trunk Seizure Option. The accumulated duration and peg count of a call will begin at trunk seizure time instead of at the time the call was established.

The Traffic Monitor output that starts during the same time that the regular traffic report starts is impacted if the Traffic Period Option is enabled. With this option enabled, the duration of all currently established calls is accumulated at the end of the traffic period. Therefore, this additional duration is also accumulated in the next minute's traffic monitor output. For example, the Traffic Monitor feature and the Traffic Period Option are both enabled. Regular traffic reports are output every half hour. The difference in the accumulated duration from 10:29 to 10:30 may increase dramatically due to the additional durations accumulated for currently established calls at the end of this traffic period.

Feature packaging

The Trunk Traffic Reporting Enhancement feature is included in base X11 system software.

Feature implementation

LD 17 – Both options can be allowed or denied on a system-wide basis using this overlay.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PARM	Release 19 gate opener.
...		
- TPO	(NO) YES	Traffic Period Option. Enter YES to enable, NO to disable, and <CR> to keep the current value.
- TSO	(NO) YES	Trunk Seizure Option. Enter YES to enable, NO to disable, and <CR> to keep the current value.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	9.32
Networking:	No

Trunk Verification from a Station

Trunk Verification from a Station (TVS) provides the capability for a classmarked 2500-type telephone (i.e., basic push-button set having no feature keys) to seize a particular trunk within a trunk group, receive a dial tone, and outpulse digits to complete a call to a remote maintenance site. This feature is used as part of a PC-based Network Management system to allow physical testing of each trunk in the network.

Any compatible, customer-provided PC-based PBX administration and maintenance system accesses the trunk to be tested and calls a remotely located customer-provided responder. The responder supplies the various tones needed to perform the trunk test. The PC then stores and processes the results. Once the testing is complete, the PC disconnects from the tested trunk and accesses the next trunk in the route.

To the system, the PC appears as a 2500-type telephone, which requires the capability to seize a particular trunk member within a trunk route.

Operating parameters

It is recommended that the telephone with a Trunk Verification Allowed (TVA) Class of Service also have CFW All Calls To External DN Denied (CXFD), CFW Busy Denied (FBD), and CFW No Answer Denied (FND) Classes of Service. This setup prevents any restricted telephone from accessing trunks by calling the TVA telephone and subsequently getting transferred or forwarded.

Also, it is strongly recommended that this unit not be configured with an LPA. This will prevent the unit from initiating the PBXT (test message waiting lamps) command in LD 32.

The telephone with a Trunk Verification Allowed (TVA) Class of Service should also be assigned Warning Tone Denied (WTD) Class of Service. This will prevent Attendant Busy Verification, which could impair the trunk frequency measurements that take place during a TVS call. This also prevents the trunk that this telephone has seized from being barged into by the attendant.

Trunk Verification from a Station is not applicable to B-channels on digital links.

When using the Trunk Verification feature to test network trunks, any trunk state other than an idle, such as busy, disabled or maintenance busy, an overflow tone is returned.

Feature interactions

The environment in which the TVS feature will be invoked is a machine environment. That is, the user of the 2500-type telephone with this feature will usually be a PC-based maintenance system. Therefore, minimal interaction exists with other features.

When the 2500-type telephone with a TVA Class of Service makes a TVS call, any Trunk Group Access Restrictions/Trunk Access Restriction Groups (TGAR/TARG) defined in the system are removed for this call.

When a trunk group is busied out by an Attendant Console, access to that trunk group is not allowed with the TVS feature.

Feature packaging

Trunk Verification from a Station (TVS) package 110 has no feature package dependencies.

Feature implementation

LD 10 – Allow or deny Trunk Verification from a 2500 telephone.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	(TVD) TVA	(Deny) allow TVS.
	DTN	Digitone service is required for 2500 telephones.

Feature operation

To verify that a trunk is working properly (from a 2500 telephone with TVA Class of Service), follow these steps:

1 Lift the handset.

2 Dial SPRE + 70 + ACOD + mmm

where:

SPRE	is the special function access prefix
70	is the special access code for the TVS feature
ACOD	is the access code of the trunk group to be tested, and
mmm	is the number of the trunk member that is to be seized; mmm must be three digits (e.g., 001).

Introduced in X11 Release:	All
Networking:	No

Uninterrupted Line Connections

Uninterrupted Line Connections are connections assigned Warning Tone Denied (WTD) Class of Service. The feature prohibits the imposition of any Camp On or intrusion tones on that line.

This feature is recommended for modem or data lines.

Operating parameters

There are no feature requirements.

Feature interactions

Attendant Barge-In
Attendant Busy Verify
Override

These features cannot be applied to stations with a WTD Class of Service.

Camp-On

A call can be camped on to a station with a WTD Class of Service, but tone is not provided.

Feature packaging

Uninterrupted Line Connections is included in base X11 system software.

Feature implementation

LD 10 – Assign Warning Tone Allowed for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	(WTA) WTD	Warning tone (allowed) denied.

LD 11 – Assign Warning Tone Allowed for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	(WTA) WTD	Warning tone (allowed) denied.

LD 14 – Assign Warning Tone Allowed for trunks.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaa	Trunk type, where: aaa = ADM, AID, ATVN, AWR, CAA, CAM, COT, CSA, DIC, DID, FEX, ISA, MDM, MUS, PAG, RAN, RCD, RLM, RLR, TIE, or WAT.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	(WTA) WTD)	Warning tone (allowed) denied.

Feature operation

No specific operating procedures are required to use this feature.

United Kingdom Analogue Hardware Support

The United Kingdom analogue Hardware Support feature provides the following capabilities:

— UK Analogue Trunk Enhancements, and

— UK Transmission Plans.

UK Analogue Trunk Enhancements

Software changes have been implemented for the following hardware packs, in order to comply with UK standards:

— XDID (Extended DID trunk card)

— XCOT (Extended Central Office trunk card)

— XTD (Extended Tone Detector card), and

— XFEM (Extended Flexible E&M trunk card).

XDID

Situation	Solution
A DID trunk is not available for a new call.	A backward signal is sent to the Public Switched Telephone Network.
A short line and long line DID trunk requires support.	A 2dB Short Line (SHL) and Long Line (LOL) pad matrix have been defined.

XCOT

Situation	Solution
Support the following types of disconnect signaling required for Central Office trunks: • Earth Signaling (Ground Start), • Loop Calling (Disconnect Clearing), and • Loop Calling (Guarded Release) signaling.	The appropriate disconnect sequences have been programmed.
For Periodic Pulse Metering (PPM), an option is required to default to a meter pulse frequency of 50 Hz (the XCOT pack for the UK can only accept this value).	In the Route Data Block, if the PPM frequency is not prompted, the value will default to 50 Hz.
For Periodic Pulse Metering, the counting of buffered and unbuffered pulses.	The software has been modified to support both buffered and unbuffered PPM pulses.
A time-configurable detector is required to monitor the disconnection of loop trunks, disconnect clear trunks, and release guard trunks.	The Loop Calling Timer (LCT), with a configurable range of 128-32640 milliseconds, has been introduced in the Route Data Block.
UK ringing must be recognized.	To recognize UK ringing, the default value of the ring validation timer has been changed from 512 milliseconds to 256 milliseconds.
UK COT with Earth Signalling (Ground Start) or Loop Calling Disconnect Clearing provides hardware answer supervision.	The software has been modified to support answer supervision for both Earth Signalling (Ground Start) and Loop Calling Disconnect Clearing. Prompt SUPN appears for both types of signaling in LD 14. Answer supervision is not provided for Loop Calling Guarded Release.

XTD

The XTD pack can be configured, on a per-call basis, for either Dual-tone Multifrequency (DTMF) or Dial Tone Detection (DTD) signaling.

XFEM

The XFEM pack supports recorded announcement trunks, paging trunks, and music trunks, two-wire E&M, four-wire E&M, and 2280 Hz TIE trunks.

UK Digital Transmission Plans

Software changes have been implemented in order to comply with UK digital transmission plans for the following:

— Digital trunks, and

— Meridian modular telephones.

Digital trunks

Situation	Solution
The transmission parameter values for digital trunks must be fixed.	The transmission parameter values for digital trunks are automatically downloaded, based on a zero default value.

Meridian modular telephones

Situation	Solution
The transmission parameter values must be fixed and automatically downloaded, on a per-system basis.	The software has been changed to prevent transmission parameter prompts from appearing. The transmission parameters will be fixed for the UK, and will be downloaded on a per-system basis.

Operating parameters

There are no feature requirements.

Feature packaging

United Kingdom Program (UK) package 190.

Feature implementation

LD 13 – Respond to the following prompts to define the DTD/DTR data block changes.

Prompt	Response	Description
...		
TYPE	XTD	Extended Dial Tone Detector and Digitone Receiver data block.
...		
XTDT	(0)-7	Extended Tone Detector Table Number, prompted when type = XTD. If a table other than 0 is entered, it must exist in LD 97.
- DTO	(NO) YES	Dial Tone Detection Only. (NO) = Do not disable DTR detection. YES = Disable DTR detection, only perform dial tone detection.

LD 14 – Respond to the following prompts to define the UK trunks, and associated signaling type and pad setting Class of Service.

Prompt	Response	Description
...		
XTRK	XFEM XDID XCOT	Extended Flexible E&M trunk card. Extended DID trunk card. Extended CO trunk card.
...		
SIGL		Trunk signaling.
	LDC	Loop calling, disconnect clear. Accepted when TYPE = COT and UK package is equipped.
	LGR	Loop calling, guarded release. Accepted when TYPE = COT and UK package is equipped.

...		
CLS	(SHL) LOL	(Short line) Long line Class of Service.
	NTC TRC VNL.	Transmission Class of Service, where: NTC = Non-transmission Compensated TRC = Transmission Compensated, and VNL = Via Net Loss.
		For E&M4 Wire and AC15 defined on XFEM trunks, NTC is used for PBX-to-PSTN Link connections, while VNL is used for PBX-to-PBX TIE connections.
		SHL replaces TRC and LOL replaces NTC and VNL for XDID and XCOT trunks in Phase 7C and later.

LD 16 – Define the Loop Calling Detection Timer, and how the ground signal from a Recorded Announcement (RAN) machine should be interpreted for XFEM cards.

Prompt	Response	Description
...		
TIMR	LCT 0-128-1280	Loop Calling Detection Timer in milliseconds.
		Default for COT trunks = 128. Default for all other trunks = 256.
...		
GRD		Determines how the ground signal from a RAN machine should be interpreted for XFEM cards.
	(PLAY)	The ground signal from the RAN machine indicates that the machine is playing.
	IDLE	The ground signal from the RAN machine indicates that the machine is idle.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	19
Networking:	No

User Selectable Call Redirection

X11 Release 19 and later includes User Selectable Call Redirection (USCR), which enhances the implementation of several existing features. First, it enables the user to modify DNs at the telephone for the following redirections:

— Flexible Call Forward No Answer DN (FDN)

— Hunt DN (HUNT)

— External Flexible Call Forward No Answer DN (EFD), and

— External Hunt DN (EHT).

The Station Control Password feature must be active, with passwords defined in LD 15, for the user to change these redirection DNs.

Second, it expands the number of selectable Ringing Cycle Options (RCOs) for Flexible Call Forward No Answer (CFNA) from one to three.

User assignment of redirection DNs

Prior to X11 Release 19, changing the redirection DNs for FDN, HUNT, EFD, and EHT required a service change to LDs 10 and 11. USCR permits the user to modify any of these four DNs from a rotary, push-button, or digital telephone.

Depending on the type of telephone, there are three ways to access this feature: using a Special Service Prefix Code (SPRE 9915), a Flexible Feature Code (FFC), or the User Selectable Redirection (USR) key.

The user can also change the RCO from a telephone after accessing USCR. For security reasons, the user must enter the Station Control Password (SCPW) before changing the redirection DNs or the RCO.

Ringing Cycle Options (RCOs) for CFNA

The original implementation of Call Forward No Answer provided a single option (CFNA in LD 15) that defined the number of normal ringing cycles before CFNA treatment. The value could be in the range of 1-15, with a default of 4. This value determined how many times the telephone rang before CFNA treatment was initiated.

The CFNA prompt is now replaced with prompts CFN0, CFN1, and CFN2, each of whose value can be in the range of 1-15, with a default of 4. The number of distinctive ringing cycles for CFNA is also expanded. The DFNA prompt in LD 15 is replaced with DFN0, DFN1, and DFN2, with the same value range and default.

Additionally, the Ringing Cycle Option (RCO) prompt appears in LD 10 and 11 for each telephone. Its value, in the range of 0-2, is a pointer to the CFNx and DFNx entries in the Customer Data Block. The following chart explains the relationship of the RCO value and the CFNx and DFNx entries in the Customer Data Block.

Table 150
Relationship between RCO value and CFNx, DFNx contents

An RCO value (per telephone) of	Selects these CFNA and DFNA entries (with sample contents shown)	And has this effect
0	**CFN0 (Default value of 4)**	CFNA treatment after four rings
	DFN0 (Value set to 2)	CFNA treatment after two distinctive rings
1	**CFN1 (Value set to 6)**	CFNA treatment after six rings
	DFN1 (Value set to 5)	CFNA treatment after five distinctive rings
2	**CFN2 (Value set to 3)**	CFNA treatment after three rings
	DFN2 (Default value of 4)	CFNA treatment after four distinctive rings

Operating parameters

To assign or print the RCO for a telephone requires that it have the Flexible Call Forward No Answer Allowed (FNA) Class of Service or Message Waiting Allowed (MWA) Class of Service.

The user's telephone must have User Selectable Redirection Allowed (USRA) Class of Service and a Station Control Password (SCPW). The user must enter the correct password to access USCR.

Basic Rate Interface (BRI) telephones do not support USCR because they cannot access SPRE or FFC, and have no feature keys. Therefore, BRI telephones will always use the entries for CFN0 and DFN0.

The user cannot use USCR to initially configure call redirection features. The features must be equipped, and the initial call redirection DNs must be established, via a service change.

This feature cannot be used remotely. A user can only change redirection DNs or the RCO for the telephone being used to access USCR.

Feature interactions

Automatic Call Distribution

An Automatic Call Distribution (ACD) DN cannot be stored as a redirection DN unless the ACD queue is defined as a Message Center.

Attendant Administration

Attendant Administration does not support assigning the USR key, RCO, or USRA/USRD Class of Service.

Autodial

USCR does not support Autodial; Autodial cannot be used to dial all or part of the digits for USCR programming.

Call Forward All Calls

When CFW redirects a call from telephone A to telephone B, and telephone B does not answer, the RCO of telephone B determines how long it rings. After the designated number of rings, the FDN of telephone A redirects the call.

Call Forward by Call Type

USCR enables a user to assign EFD from the telephone.

Call Forward No Answer
Flexible Call Forward No Answer

In X11 Release 19 and later, the single parameters previously used to define normal ringing cycles (CFNA) and distinctive ringing cycles (DFNA) are expanded to three (CFN0-2 and DFN0-2), with the Ringing Cycle Options (RCO) parameter used to select the specific CFNA and DFNA entries for each telephone.

Call Forward No Answer, Second Level

The number of ringing cycles before Second Level Call Forward No Answer (SFA) is determined by the RCO for the ringing DN, as with CFNA.

Call Redirection by Time of Day

User Selectable Call Redirection is not supported by Call Redirection by Time of Day.

Dial Access to Features and Services

The 9915 feature code accesses USCR from an analog (500/2500 type) telephone or a Meridian 1 proprietary telephone. The user dials this code after dialing the SPRE.

Directory Number Delayed Ringing

With User Selectable Call Redirection (USCR) a user can change the number of CFNA/DFNA ringing cycles. If the user changes the CFNA/DFNA value so that CFNA takes place before the Directory Number Delayed Ringing timer runs out, none of the SCN/MCN keys will receive an audible notification.

Distinctive/New Distinctive Ringing

The single parameter previously used to define distinctive ringing cycles (DFNA) is expanded to three (DFN0-2), with the Ringing Cycle Options (RCO) parameter used to select the specific DFNA entry for each telephone.

DPNSS1 Diversion

The User Selectable Call Redirection feature triggers Diversion Validation. If the numbering plan is DPNSS1 then diversion occurs. Numbering plan routes are checked to determine if redirection DN's are through DPNSS1 on a first choice route basis. If the number plan is not a DN through DPNSS1, then User Selectable Call Redirection works as usual.

Enhanced Hot Line
Flexible Hot Line

An analog (500/2500 type) telephone with a Hot Line feature cannot use User Selectable Call Redirection, because it cannot access any features through SPRE or FFC.

Hunting

User Selectable Call Redirection permits a user to change the HUNT DN or EHT from a telephone. An attendant DN is only allowed for HUNT and EHT if the customer has the attendant defined as a message center (LD 15 – MATT=YES).

Message Center (MC) and Message Waiting

USCR affects the number of times the DN rings before the call is forwarded to the Message Center. The RCO in the Terminal Number (TN) block of the Multiple Appearance Redirection Prime (MARP) for the called DN determines the number of times the DN rings.

Multiple Appearance Redirection Prime (MARP)

When a Multiple Appearance DN is rung, the determination of the number of ringing cycles for CFNA depends on the value of the MARP prompt in LD 17. If the value is "YES," the number of ringing cycles is determined by the RCO number of the DN that is classified as a MARP TN. If the DN is a Multiple Appearance DN (MADN), the RCO values in the other TN blocks for that DN are ignored.

If the MARP value is "NO," the RCO is taken from the first TN in the DN block with a primary appearance of the DN. If there is none, the last TN in the DN block is used.

Pretranslation

If Pretranslation (package 92) is enabled, the digits entered as the redirection DN are pretranslated before they are stored. Note that no Pretranslation occurs when the redirection DNs are used in such call processing features as Hunting or CFNA, eliminating the possibility that the redirection DN is pretranslated twice.

Short Hunting

USCR does not support changing the HUNT or EHT for a telephone with Short Hunt enabled. USCR also does not support entering “000” from a telephone as the HUNT.

Speed Call

Speed Call is not supported by USCR.

Feature packaging

User Selectable Call Redirection is available as part of X11 Release 19. Flexible Feature Codes (FFC) package 139 is a prerequisite for the user activation part of this feature because it provides for the Station Control Password.

Feature implementation

Responses to the LD prompts shown in the following tables set up USCR. Responses differ depending on the type of telephone and the type of access being set up.

LD 15 – Setting up USCR in the Customer Data Block.

Prompt	Response	Description
REQ	NEW CHG	ADD, or change.
TYPE	CDB RDR	Customer Data Block. Release 21 gate opener.
CUST	0-99 0-31	Customer number. For Option 11C.
- CFN0	1-(4)-15	Number of normal rings for CFNA, Option 0.
- CFN1	1-(4)-15	Number of normal rings for CFNA, Option 1.
- CFN2	1-(4)-15	Number of normal rings for CFNA, Option 2.
- DFN0	1-(4)-15	Number of distinctive rings for DFNA, Option 0.
- DFN1	1-(4)-15	Number of distinctive rings for DFNA, Option 1.
- DFN2	1-(4)-15	Number of distinctive rings for DFNA, Option 2.
TYPE	FFC	Release 21 gate opener.
...		
- SCPL	(0-8	Length of Station Control Password. If 0 = password disabled, USCR cannot be used.

LD 10 – Setting up USCR for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	500	Telephone type.
RCO	(0) 1 2	Ringing Cycle Option for CFNA, in the range of 0-2, with a default of 0.
SCPW	xxx...xx	Station Control Password.
CLS	(USRD) USRA	User Selectable Redirection Class of Service (permitting SPRE and FFC access) (denied) allowed.
Note: The technician can use easy change to change the RCO and USRA/USRD CLS. At the ITEM prompt, type RCO <value> where the value is 0-2.		

LD 11 – Setting up USCR for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
RCO	(0) 1 2	Ringing Cycle Option for CFNA, in the range of 0-2, with a default of 0.
SCPW	xxx...xx	Station Control Password.
CLS	(USRD) USRA	User Selectable Redirection Class of Service (permitting SPRE, FFC, and USR key access) (denied) allowed.
KEY	xx USR	Key number of the USR key.
Note: The technician can use easy change to change the RCO and USRA/USRD CLS. At the ITEM prompt, type RCO <value> where the value is 0-2.		

LD 57 – Setting up USCR.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
CUST	0-99 0-31	Customer number. For Option 11C.
CODE	USCR ALL	Prompt for USCR FFC, or all FFC code types.
USCR	xxxxxxx	USCR FFC (1-7 digits).
	yyyyyyy	Define additional FFC codes, as needed.
	<CR>	Ends the entry of FFC codes.

Feature operation

As a prerequisite to accessing the feature, the conditions shown in Table 151 must be met for the selected access method.

Table 151
Requirements for accessing USCR

Requirement	Access Method		
	USR Key	SPRE	FFC
FFC package equipped	Yes	Yes	Yes
SCPL is defined (>0)	Yes	Yes	Yes
SCPW is defined	Yes	Yes	Yes
Telephone has USR key	Yes	No	No
USRA Class of Service defined	Yes	Yes	Yes
SPRE defined	No	Yes	Yes
USCR FFC defined	No	No	Yes

To assign/query a redirection DN using SPRE:

1. Take the telephone off-hook, or press the DN key on a digital telephone.
2. Enter the SPRE.
3. Enter the USCR feature access code (9915).
4. Enter the Station Control Password.
5. Enter the USCR option code, as shown in Table 152.

Table 152
USCR option codes

Code	Used to assign
1	FDN redirection DN
2	HUNT redirection DN
3	EFD redirection DN
4	EHT redirection DN
5	RCO

6. Enter the new RCO if assigning the RCO; enter the redirection DN if assigning the DN.
7. Place telephone on-hook, or press the Rls key on a Meridian 1 proprietary telephone.

To assign or query a redirection DN using the USR key:

1. Press the dark USR key.
2. Enter the Station Control Password.
3. Enter the USCR option code from Table 152.
4. Enter the new RCO if assigning the RCO; enter the redirection DN if assigning the DN.
5. Press the USR key again.

To assign or query a redirection DN using an FFC:

1. Take the telephone off-hook, or press the DN key on a Meridian 1 proprietary telephone.
2. Enter the USCR FFC.
3. Enter the Station Control Password.
4. Enter the USCR option code, as shown in Table 152.
5. Enter the new RCO if assigning the RCO; enter the redirection DN if assigning the DN.
6. Place telephone on-hook, or press the **Rls** key on a Meridian 1 proprietary telephone.

Variable Flash Timing and Ground Button

These two methods of operation allow an analog (500/2500 type) telephone user to obtain special dial tone and activate various system features while on an established call. They are also used to return to the original call. Both of these functions are referred to as a recall. The following are the two parts of the feature:

Variable Flash Timing

This part is an enhancement to Flash Timing. It allows further flexibility in defining the limits for the flash. A minimum range of 20 to 768 milliseconds has been added and the maximum range has been extended to 1500 milliseconds. These settings are made on a customer basis in LD 15. A switchhook flash of less that the minimum is ignored and one of greater than the maximum is read as a disconnect. All flashes between the minimum and maximum provide a recall.

Ground Button

This part is an alternative to Flash Timing. It requires the installation of a ground button line card in place of a regular 500-type line card and analog (500/2500 type) telephones which have the ground button capability. The ground button can be depressed for any length of time over the minimum flash timing to provide a recall.

Operating parameters

Variable Flash Timing and Ground Button Operation are supported only on Digitone sets.

Ground Button Operation requires that a QPC532 Ground Button line card be installed on sets that have the capability, rather than a regular 500-type line card.

Using SPRE codes, it is possible to invoke the same features from an analog (500/2500 type) telephone as from a feature telephone.

Feature interactions

Message Waiting

The Ground Button Recall message to the software uses the same data store as the Message Waiting feature. The telephone state indicates which feature is active. The state is idle for Message Waiting and active for Ground Button Recall.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 15 – Modify data for each customer member to be configured and assign the minimum and maximum flash time.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	CDB TIM	Customer Data Block. Release 21 gate opener.
...		
- FLSH	xxx yyy	Minimum and maximum switchhook flash timer in milliseconds. xxx = 20-(45)-768. yyy = 384-(896)-1500. The timing specified will be used for EPE equipment only. XPE equipment will use the FLSH specified in LD 97.

Feature operation

Variable Flash Timing

Any switchhook flash between 20 and 1500 milliseconds provides a recall.

Ground Button

Pressing the ground button for any length of time over 20 milliseconds provides a recall.

Variable Guard Timing

The guard timing capability for a trunk prevents outgoing calls from reseizing trunks for a specified time after disconnection, thereby protecting trunks against glare conditions. This feature allows the customer to specify one guard timing interval for incoming call disconnection and one guard timing interval for outgoing call disconnection.

Operating parameters

There are no feature requirements.

Feature interactions

There are no interactions with other features.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Create or modify the data blocks for trunk routes.

Prompt	Response	Description
...		
CFWR	(NO) YES	CFW restriction (not allowed) allowed.
IDOP	(NO) YES	Respond YES to allow the trunk CDR for internal calls to identify the originating station instead of the forwarding station.
TIMR	GTI 128-(896)-32640	Incoming Guard timer.
TIMR	GTO 128-(896)-32640	Outgoing Guard timer.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	1
Networking:	No

Voice Call

Voice Call allows you to talk through the speaker of a Meridian digital telephone from another Meridian digital telephone. The called party does not have to lift the handset to hear you. For a two-way conversation, the called party must lift the handset or activate Handsfree, unless Handsfree Voice Call is enabled.

If the called telephone is busy on another DN, the caller hears continuous ringing. The called party hears a single beep and the Voice Call DN key flashes. If the telephone is busy on the Voice Call DN, the caller hears a busy tone. A fast busy tone may indicate that the Voice Call DN is no longer available (it may not be a Single Appearance DN).

Handsfree Voice Call

Handsfree Voice Call is an X11 Release 19 system feature that can be used with such telephones as the M2112, M2317, and M2616.

Handsfree Voice Call provides the option of configuring VCC/DIG (with voice option) to be answered in either Handsfree mode or loudspeaker only mode. Calls answered in Handsfree (HVA) mode establish a two-way voice path, while those answered in loudspeaker only (HVD) mode establish a one-way voice path from the calling telephone to the destination telephone.

Operating parameters

Both telephones must be Meridian digital telephones.

The Voice Call DN must be single appearance.

Handsfree Voice Call allowed/denied is set at the system level and can only be used with digital telephones that have Handsfree capabilities (such as the M2112, M2317, M2616). It requires Handsfree Allowed/HFA Class of Service on the destination telephone, which is set at the telephone level. Basic Rate Interface (BRI), M3000, and SL-1 telephones do not support the Handsfree feature.

Feature interactions

Auto Answer Back

This feature is not affected by the Handsfree Voice Call feature.

Automatic Line Selection

This feature is not selected by automatic Outgoing Line Selection. It is selected for Incoming Ringing and Non-Ringing Line Selection.

Call Party Name Display

The telephone originating a Voice Call displays the called DN's Call Party Name Display. The called telephone shows the caller's DN and name on its display.

Display of Calling Party Denied

Display information on sets involved in a Voice call is based on the individual Class of Service of each set.

Flexible Feature Code Boss Secretarial Filtering

A call to a Voice Call key on a boss set with filtering active is not filtered to the secretary set.

Flexible Voice/Data Terminal Number

If a dynamic TN has a single appearance DN key that terminates on a Voice Call (VCC) key, the called party hears a single beep if occupied on another DN. However, if the called party is a dynamic TN in data mode, the DN key lamp flashes. A beep is not provided.

Hot Line

The terminating DN of a Voice Call arrangement may be the incoming DN of a two-way Hot Line.

When engineering call-modification paths (such as Hunting and Call Forward No Answer), the Hot Line Restriction option will cancel the normal call-modification operation for internal non-Hot Line calls.

Manual Signaling

The same DN can be used for both Voice Call and Manual Signaling (Buzz) as long as it remains a Single Appearance DN.

Multiple Appearance DNs

If a Voice Call DN is added to a second telephone, the DN becomes a Multiple Appearance DN (MADN). Voice Call no longer works on that DN and fast busy tone is returned.

On Hold on Loudspeaker

It is possible to program this feature with a loudspeaker DN, but operation will be the same as for direct dial to a loudspeaker DN.

Feature packaging

Voice Call requires the Optional Features (OPTF) package 1.

Handsfree Voice Call requires X11 Release 19 or later.

Feature implementation

LD 11 – Add or change Voice Call for the originating Meridian 1 proprietary telephone.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number. For Option 11C.
KEY	xx SCR yyy...y	Adds a single appearance single call key on the terminating telephone, where: xx = key number, and yyy...y = the DN assigned to the Voice Call key for the originating telephone.
KEY	xx VCC yyy...y	Adds a Voice Call key on the originating telephone, where: xx = key number, and yyy...y = the DN of the terminating telephone. This key activates the feature.

LD 15 – Add or change Handsfree Voice Call for the Meridian 1 system.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CDB FTR	Customer Data Block. Release 21 gate opener.
CUST	0-99 0-31	Customer number. For Option 11C.
- OPT	(HVD) HVA	Handsfree Voice Call (denied) allowed.

Feature operation

Voice Call

To make a Voice Call:

— Lift the handset and press **Voice Call**. The DN is automatically dialed. If the called telephone is busy on another DN, you hear continuous ringing. If the telephone is busy on the Voice Call DN, you hear busy tone.

To end a Voice Call:

— Press **Rls**.

To answer a Voice Call on an idle telephone:

— Let the call ring once. The call is answered automatically, activating the Voice Call DN over the speaker. For a two-way conversation, lift the handset.

If busy on another DN, you hear a single beep and the Voice Call DN flashes. You must end your present call to receive the Voice Call.

Handsfree Voice Call

HVA option

The originating telephone (telephone A) places a VCC/DIG call to the destination telephone (telephone B).

1 Telephone B rings once.

2 After one ring, telephone B automatically answers the call in Handsfree mode.

The DN and Handsfree LCDs are lit and a two-way voice path is established.

HVD option

Telephone A places a call to telephone B.

1 Telephone B rings once.

2 After one ring, telephone B automatically answers the call in loudspeaker only mode.

The DN LCD is lit and the Handsfree LCD remains dark, establishing a one-way voice path from telephone A to telephone B. At this point, telephone A is unable to hear the person at telephone B.

To reestablish a two-way voice path, telephone B must either go off-hook or press the Handsfree button.

Note: Busy calls are not changed by Handsfree Voice Call.

Introduced in X11 Release:	15.58F
Networking:	Yes

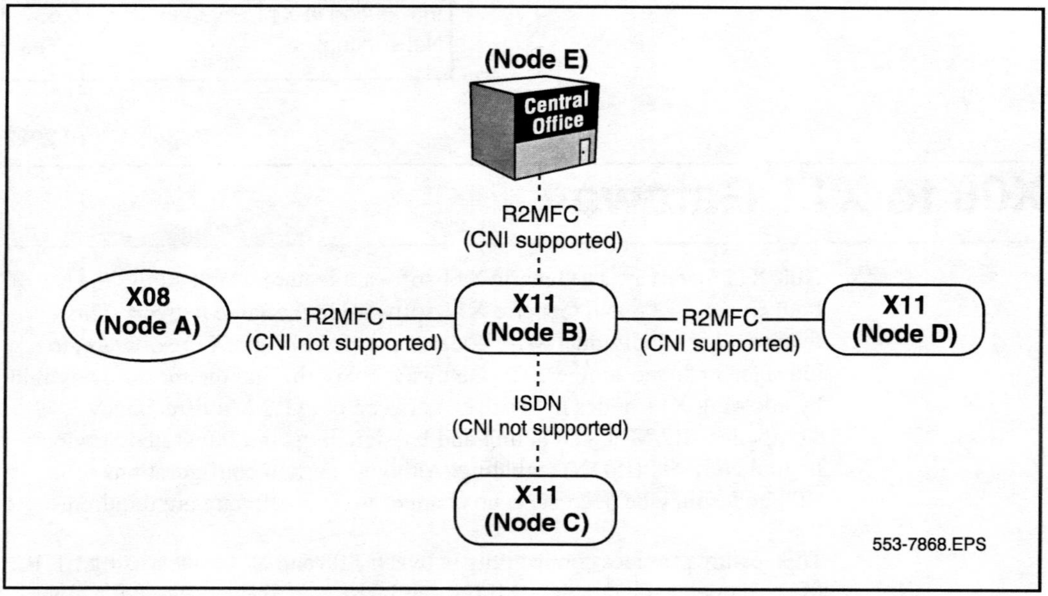
X08 to X11 Gateway

X08/X11 Gateway is a Generic X11 software feature which allows the use of both Generic X08 and Generic X11 software in the same network.This feature allows individual SL-1 nodes, running X08 and X11 software, to interface with one another. The Gateway makes this interconnection possible by allowing X11 nodes to "bridge" between both R2 Multifrequency Compelled (R2/MFC) signaling and L1 signaling, and Integrated Services Digital Network (ISDN) signaling. Although certain configurations of the X08 nodes may be necessary, no changes to X08 software are required.

This feature provides connectivity between X08 and X11 nodes, using L1, R2 Multifrequency Compelled (MFC), and Integrated Services Digital Network (ISDN) signaling protocols. The X08 L1 Signaling supports call setup, a numbering plan and Calling Number Identification (CNI). However, the L1 Signaling that is provided into X11 is a subset of the X08 L1 Signaling, supporting only the supplementary services required to support CNI and the suppression of Bring Up Receiver (BUR) signals.

Figure 91 summarizes the types of R2 MFC connections and tandems that are supported by the X08 to X11 Gateway.

Figure 91
R2 MFC Connections and Tandems



X08 to X11 connections using R2 MFC routes (Node A to Node B) – CNI is not supported because X08 does not provide outgoing CNI signaling.

X11 to a private exchange using R2 MFC routes (Node B to Node E) – CNI is supported in both directions (DID/DOD).

Tandems using R2 MFC and ISDN routes, as follows:

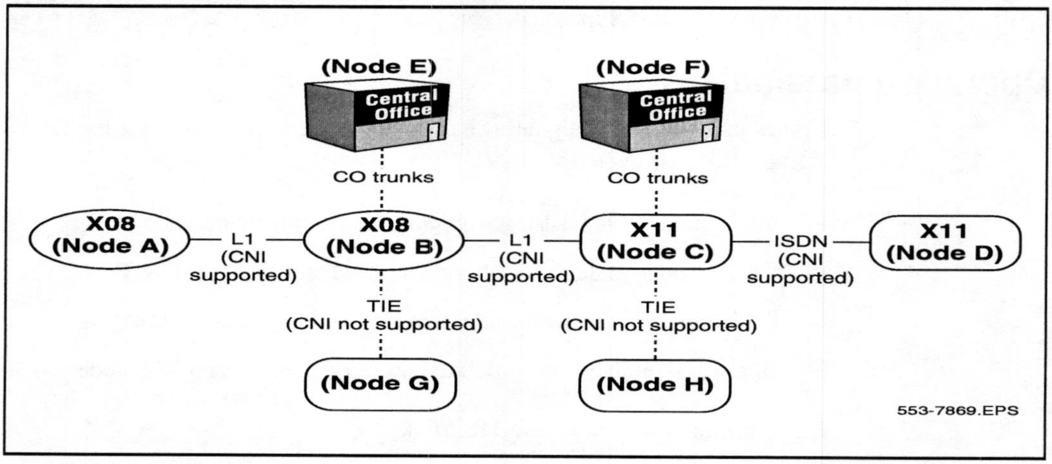
— Tandems from an X08 node to an X11 node using R2 MFC routes to another X11 node using ISDN routes (Node A to Node B to Node C). CNI is not supported for this tandem.

— Tandems from one X11 node to another X11 node using R2 MFC routes to another X11 node using ISDN routes (Node D to Node B to Node C). CNI is not supported for this tandem.

— Tandems from a Public Exchange to an X11 node using R2 MFC CO routes to another X11 node using ISDN routes (Node E to Node B to Node C).

Figure 92 summarizes the types of L1 connections and tandems that are supported by the X08 to X11 Gateway (tandemning to X11 nodes using the R2 MFC Signaling is not allowed):

Figure 92
L1 MFC Connections and Tandems



X08 to X11 connections using L1 routes (Node B to Node C) – CNI is supported.

X08 or X11 connections to a private exchange using CO routes (Node B to Node E or Node C to Node F) – These routes can be analog or digital, and are non-R2MFC. CNI is not supported.

X08 or X11 connections to a private exchange using TIE routes (Node B to Node G or Node C to Node H) – These routes can be analog or digital, and are non-R2MFC. CNI is not supported.

Tandems using L1 and ISDN routes, as follows:

— Tandems from an X08 node to an X11 node using L1 routes to another X11 node using ISDN routes (Node B to Node C to Node D). CNI is supported.

— Tandems from an X08 node to an X11 node using L1 routes to a private exchange using analog or digital routes (Node B to Node C to Node F). CNI is not supported for this tandem.

— Tandems between X08 and X11 nodes using L1 routes to a node using TIE routes (Node B to Node C to Node H, and Node C to Node B to Node G). CNI is not supported.

Operating parameters

Routes using R2/MFC signaling can only be tandemed to routes using L1 signaling in cases where:

— the L1 route uses L1 Basic signaling (no supplementary services);

— the X11 node makes no Calling Number Identification (CNI) requests;

— the X08 node makes no call extensions for Ring Again (RGA); and

— signal assignment is co-ordinated between the X11 and X08 nodes.

L1 signaling in X11 must use TIE trunks.

L1-signaled routes will support CNI only when End-to-End Signaling is used.

The following groups of features do not operate on L1-signaled calls between X08 and X11 nodes:

— features requiring Bring Up Receiver (BUR) signals;

— call diversions;

— X08 trunk optimization;

— call transfer to an unestablished connection;

— Break-in, Recall, Incoming Call Identification (ICI) requests and Night Service Notification attendant features; and

— Ring Again (RGA).

X08 L1 signaling allows only one unsupervised trunk in a call connection. An X11 node tandeming an L1 connection from an X08 node does not inform the X08 node of unsupervised-trunk usage.

R2/MFC tandems support End-to-End Signaling only when the tandem node uses either the same R2/MFC table for both trunks or uses two tables with identical contents and the same End-to-End Signaling code. Calling Number Identification (CNI) is carried end-to -end even where End-to-End Signaling is not available.

X08-to-X11 connections, using R2/MFC, do not support CNI. Outgoing CNI, on a tandem R2/MFC connection from an X08 node, uses the customer identifier of the tandeming X11 node, plus the Access Code of the route from the X08 node.

CNI is not supported over R2/MFC-to-ISDN tandem connections.

A third level of R2/MFC signaling, consisting only of backward signals, is not supported. This level of signaling is used for coin-box calls or calls from subscribers with home meters.

CNI in Call Detail Recording (CDR) records will have the same length only when all DNs, route access codes, trunk identifiers and attendant identifiers have the same length.

X08 does not have Integrated Services Digital Network (ISDN) capabilities.

Feature interactions

The network supported features using Gateway depends on the specific types of connection involved in any particular call.

Calling Number Identification (CNI)

Calling Number Identification (CNI) is supported on R2/MFC signaling connections between X11 nodes and Central Offices (COs), in both directions of calling, provided that the trunk being used has CNI-allowed Class of Service. CNI has the following characteristics across this type of connection:

- CNI begins with an optional customer identifier, 1-8 digits long;
- the customer identifier is followed by a caller identifier (a DN of 0-7 digits, an attendant identifier, a trunk identifier or a route access code);
- the attendant identifier has a maximum of 4 digits (identified on a customer basis); if the attendant identifier has not been defined, the attendant DN is used;

- the trunk identified has 0-7 digits (as assigned in Overlay 14)
- the trunk identifier does not have a unique value;
- the route access code is used if the trunk identifier has not been defined;
- a maximum of 16 digits of CNI can be carried across an R2/MFC connection; and
- end-to-end CNI to the CO works when the call tandems across more than one X11 node, using R2/MFC.

CNI is not supported on tandem connections between R2/MFC and ISDN routes.

In R2/MFC connections between X08 and X11 nodes, end-to-end CNI is only supported in cases where it has been requested by the X08 node. The X08 node will not support outgoing CNI. On an outgoing connection, the CNI supplied to the far end is that of the tandeming node when the tandeming node has not received CNI on an R2/MFC connection.

CNI is fully supported on R2/MFC connections between X11 nodes. End-to-end CNI to a CO works when the call tandems across more than one X11 node, using R2/MFC. If a tandeming node does not receive CNI, that node sends its own CNI forward.

CNI is supported on L1-signaled routes only when End-to-End Signaling is used.

Network Ring Again

Network Ring Again is not supported across any R2/MFC signaling connection or across L1-signaled connections between X08 and X11 nodes.

Feature packaging

The following packages are required for X08/X11 Gateway:

For R2/MFC Signaling, the following package is required:

- R2/MFC package 128

For L1 Signaling, the following packages are required:

— L1 package 188 and

— R2/MFC package 128.

For R2/MFC—ISDN Gateway, the following packages are required:

— Integrated Services Digital Network (ISDN) package 145;

— Primary Rate Access (PRA) package 146;

— DID-to-Network package 161; and

— R2/MFC package 128.

For L1—ISDN Gateway, the following packages are required:

— R2/MFC package 128;

— L1 package 188;

— Integrated Services Digital Network (ISDN) package 145;

— Primary Rate Access (PRA) package 146;

— DID-to-Network package 161; and

— Network Attendant Service (NAS) package 159 is required when the L1—ISDN Gateway must transport CNI.

Feature implementation

LD 14–Define trunk.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	TIE	TIE trunk.
TN	l s c u c u	Terminal Number. For Option 11.
CUST	xx	Customer Number.
TKID	nnnnnnn	Trunk Type Identifier (Does not have to be unique).

LD 16–Define trunk routes.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer Number.
ROUT	0-511 0-127	Route Number. For Option 11C.
TKTP	TIE	TIE trunk type.
CCNI	(NO) YES	Call Number Indicator or CNI enabled on route.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	10.10B
Networking:	Yes

2 Mbps Digital Trunk Interface

The 2 Mbps Digital Trunk Interface (DTI2) feature provides digital connectivity between a Meridian 1 digital network loop and an external digital carrier termination. It provides digital speech on up to 30 channels at 2 Mbps on one Meridian 1 loop and the bipolar carrier terminal. Within the Meridian 1, the DTI2 operates as a general purpose sender and receiver of ABCD (signaling) bits. The DTI software sets the ABCD bits to represent the appropriate signaling for the trunk being supported.

The QPC775 clock controller allows DTI2 to be used on ST, NT, XT, MS, network enhanced XN and N, or network enhanced N half group machines.

The QPC775 clock controller allows DTI2 to be used on System Options 21, 51, 61, 71.

Refer to the 2 Mbps Digital Trunk Interface series of Northern Telecom Publications (NTPs) (553-2911-100 to 553-2911-510) for more information on this feature.

Operating parameters

There are no feature requirements.

Feature interactions

Periodic Pulse Metering

Periodic Pulse Metering operates the same for 2 Mbps DTI as for analog trunks.

Pulsed E&M DTI2 Signaling

Pulsed E&M DTI2 signaling is based on 2 Mbps DTI.

Feature packaging

2 Mbps Digital Trunk Interface (DTI2) package 129.

Feature implementation

LD 14 – Create or modify trunk data blocks for DTI2 on a per trunk basis.

Prompt	Response	Description
...		
SICA	(1)-16	Signaling Category table number. The category must already be defined in LD 73. Default is 16 if loop type = Japanese Digital Multiplex Interface (JDMI).
PDCA	(1)-16	Pad Category table number. The PAD category must already be defined in LD 73. Default is 16 if loop type = JDMI.
PCML	MU A	Indicate whether MU-Law or A -Law Pulse Code Modulation (PCM) for voice calls is active in the channel. Not prompted for JDMI loops.

LD 16 – Create or modify DTI2 trunk route data blocks.

Prompt	Response	Description
...		
DTRK	(NO) YES	Digital trunk route.
DGTP		Digital trunk type.
	(DTI) PRI DTI2 PRI2 JDMI	1.5 Mbps DTI. 1.5 Mbps Primary Rate Interface. 2 Mbps DTI. 2 Mbps Primary Rate Interface. Japanese Digital Multiplex Interface. Prompted when the DTI2 or PRI2 package is equipped.

LD 17 – Modify the system hardware and software parameters to enable or disable the feature.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	CFN CEQU	Configuration Record. Release 19 gate opener.
...		
- DTI2	0-159	2 Mbps Digital Trunk Interface (DTI) loop number. Prompted the when DTI2 or PRI2 package is equipped.

LD 73 – Implement the system hardware and software parameters to enable or disable the DTI feature.

Prompt	Response	Description
...		
TYPE	DTI2	2 Mbps DTI.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	10.10C
Networking:	Yes

2 Mbps Digital Trunk Interface Enhancements

The following enhancements have been added to the existing 2 Mbps Digital Trunk Interface (DTI2) in order to meet various customer requirements.

Alarm Handling on Direct Inward Dialing Channels

If an alarm condition occurs on a Direct Inward Dialing (DID) channel, this enhancement delays the sending of connect and disconnect signals, until the alarm condition is cleared.

Alarm Handling on Incoming Public Exchange/Central Office or Direct Inward Dialing Trunks

This enhancement clears non-established calls on incoming Public Exchange/Central Office (CO) or Direct Inward Dialing (DID) trunks when an alarm condition occurs. When the alarm condition is cleared, the calls are diverted to the attendant.

Call Clearance

This enhancement affects the handling of incoming and outgoing call clearance for Central Office (CO) calls.

Call Clearance is handled differently if the clear forward signal (CLRF) is defined, or if the clear forward signal and the IDLE signal do not have the same definition. The Call Clearance is also handled differently for outgoing and incoming calls.

For outgoing calls being disconnected by the Meridian 1 system, a clear forward and then an IDLE signal is sent by the system. The call state determines when the IDLE signal is sent. If the call is answered, the IDLE signal is sent within 300 milliseconds of the reception of a clear back signal from the CO. If the outgoing call is not answered, the IDLE signal is sent after 800 milliseconds (plus or minus 50 milliseconds) of the clear forward signal being sent. If the CO answers during this 800 milliseconds period, the Meridian 1 system continues to send the clear forward signal until it receives a clear back signal from the CO.

For outgoing calls being disconnected by the CO, a clear back signal is sent by the CO when it wishes to disconnect. The Meridian 1 system then sends a clear forward signal within 300 milliseconds of having received the clear back signal, followed by an IDLE signal within 800 milliseconds (plus or minus 50 milliseconds) of having sent the clear forward signal.

For incoming calls being disconnected by the Meridian 1 system, a clear back signal is sent by the system. Upon receiving a clear forward signal from the CO, the system sends an IDLE signal within 300 milliseconds of having received the clear forward signal.

For incoming calls being disconnected by the CO, a clear forward signal is sent by the CO when it wishes to disconnect. If the call is answered, the Meridian 1 system sends a clear back signal within 300 milliseconds of having received the clear back signal from the CO, and then an IDLE signal after 800 milliseconds (plus or minus 50 milliseconds) of having sent the clear forward signal. If the call is not answered, the system sends an IDLE signal within 300 milliseconds of having received the clear forward signal from the CO.

If an alarm condition occurs while a clear forward or clear back signal is being sent for the 800 milliseconds time period, the Meridian 1 system continues to send the signal until the alarm condition clears.

Clock Synchronization

This enhancement affects the clock synchronization controller. If a DTI loop enters its most severe alarm state (the No-New-Calls state), the Meridian 1 system disables the clock port.

Direct Inward Dialing Call Offering

The Central Office (CO) operator will be able to offer a Direct Inward Dialing (DID) call to the attendant. When a DID call terminates on a busy station, and the End of Selection Busy (EOSB) signal has been sent to the CO by the analog (500/2500 type) telephone, the CO can then send an Operator Pulse Signal (OPRS) back to the analog (500/2500 type) telephone. This OPRS causes the analog (500/2500 type) telephone to forward the call on to the attendant.

Disable Out-of-service Alarm State

This enhancement allows the Meridian 1 system to disable the Out-of-service (OOS) alarm state for an error, leaving the No New Call alarm state as the most severe state. This is done by setting the OOS threshold time for an error to zero.

Fault Signal

On an incoming call, if a Fault (FALT) Signal is received by the PBX while in an IDLE state, the PBX will respond with a Fault Signal until the CO returns to the IDLE state. On an outgoing call, the PBX will enter the FALT state if a Release Control (RCTL) signal is not received within 30 seconds.

Incoming Seizure

This enhancement, applied on a group basis, allows the Central Office to initiate a call from a lockout or far-end fault state.

Outpulsing Delay

This enhancement provides a delay before outpulsing on 2 Mbps DTI trunks.

Release Control

The PBX will now be able to send and receive the Release Control (RCTL) signal, which is sent by the called party on both incoming and outgoing calls to indicate disconnection is complete. The RCTL signal is sent by either the CO or PBX in response to a Release Clear Forward signal.

Signal Recognition

This enhancement gives the Meridian 1 system more flexibility in handling receive signals. The system can recognize a signal based on the ABCD signaling bits. Any non-significant signaling bits of a receive signal can be flagged as do-not-care. The system can then ignore these do-not-care bits before trying to determine which signal it has received.

64 Kbit Alarm Indication Signal Handling

This enhancement adds the 64 Kbit Alarm Indication Signal (AIS) as a sixth group II error state. This error state is treated the same as the other group II error states.

Centre National d'Etudes des Télécommunications enhancement for trunks entering an alarm state

This enhancement requires the QPC915 and ensures compliance to Centre National d'Etudes des Télécommunications (CNET) requirements for trunks entering an alarm state.

Trunks entering an alarm state are processed according to the type of trunk they are configured as and their previous state.

For all cases, signaling will not occur on the trunk while it is in an alarm state.

Idle trunk

When an idle trunk enters an alarm state, it will not send the "FAULT" signal.

DID trunk

Trunk seized and receiving digits

The call is taken down and the trunk is idled.

Call initiated but not answered

A timer is started when the alarm state is entered, its duration is between 20 and 40 seconds, and the called set continues to ring. During this time one of three cases may occur:

— **The timer expires:** the call is disconnected, all resources but the incoming trunk are released (delayed disconnect). This occurs even if the following case has already happened.

- **The called set answers:** no affect on the timer; the delayed disconnect will occur if the alarm is not cleared.
- **The alarm stops:** no affect on the connection, the timer is stopped and reset, and delayed signals are sent to the far end.

Call answered

The call is not dropped upon entering an alarm state. If the near-end party goes on-hook during alarm, the party is released and all resources are idled except the trunk, which is put in delayed disconnect state.

Disconnect

The alarm is ignored with respect to internal system processing, and the trunk is put in delayed disconnect state.

Outgoing Central Office Trunk (COT) call

If the destination has not answered, no action is taken when entering an alarm state. If the originator goes on-hook during an alarm state, the disconnect signal is delayed.

If the destination goes on-hook while in an alarm state, the software waits for the originator to go on-hook also. If the alarm is still present when the originator goes on-hook, system resources are idled, but the trunk is left in delayed disconnect state.

Incoming COT call

Call initiated

When entering an alarm state, the call is disconnected and all system resources are idled, including the trunk itself.

If the Attendant or Night set answered before the trunk entered the alarm state, the call is connected and the "CONNECT" signal is delayed until the alarm state is cleared.

Disconnect

The system completes the disconnect and idles the trunk without waiting for an "IDLE" signal from the far end.

Feature implementation

LD 73 – This overlay is modified to allow the implementation of the CNET enhancement for trunks entering an alarm state. The enhancement is implemented by responding YES to the new FRFW prompt in LD 73.

Prompt	Response	Description
REQ	...	
...		
GP2	...	
FRFW	(NO) YES	French Firmware. Enter YES to enable the CNET enhancement for trunks entering an alarm state processing capabilities. Requires that QPC915 packs be equipped. Enter NO if the CNET enhancement for trunks entering an alarm state processing capabilities are not required. Default is NO.

Centre National d'Etudes des Télécommunications enhancement for trunk packs exiting an alarm state

This enhancement requires the QPC915 and ensures compliance to Centre National d'Etudes des Télécommunications (CNET) requirements for packs exiting an alarm state.

At the end of a group I alarm state, the software requires the pack to send the ABCD status of each configured trunk. At the end of a group II alarm state, the software receives a report of valid ABCD status after having received a confirmation from the firmware that the firmware is functioning as expected. The system software state is updated according to this report.

Processing of overload conditions

Several enhancements occur:

— When receiving more than 100 messages per second from a 2 Mbps Digital Trunk Interface (QPC915) pack, the system attempts to go into No New Call (NNC) state and disables the error reporting. A DTA320 message is printed on the Maintenance Terminal to inform the technician. After at least two seconds have elapsed, the error reporting is re-enabled and a DTA321 message is printed. If this situation repeats itself more than 20 times within the next two minutes, the pack is disabled.

— The software status is updated to reflect the firmware status after overload.

— The overload process is able to recognize the channel causing the overload when the case arises.

Feature implementation

LD 73 – This overlay is modified to allow the implementation of the CNET enhancement for trunk packs exiting an alarm state. The enlacement is implemented by responding YES to the new FRFW prompt in LD 73.

Prompt	Response	Description
REQ	...	
...		
GP2	...	
FRFW	(NO) YES	French Firmware. Enter YES to enable the CNET enhancement for trunk packs exiting an alarm state processing. Requires that QPC915 packs be equipped. Enter NO if the CNET enhancement for trunk packs exiting an alarm state processing is not required. Default is NO.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	15.58F
Networking:	No

2 Mbps Remote Peripheral Equipment

This feature increases the serving area of a Meridian 1 machine. The 2 Mbps Remote Peripheral Equipment (RPE) feature extends the range of Meridian 1 network loops from 15 meters to any economically viable distance by transmitting the loop's control and voice signals over multiplexed pulse code modulation (PCM) connections (digital repeater equipped) in a format suitable for 2 Mbps PCM lines following CCITT recommendation G.732. This allows peripheral equipment to be placed in closer proximity to the stations served and to allow remote stations the same set of call features as local stations.

Each RPE group consists of up to four extended network loops serving the same physical site. The corresponding hardware unit is the RPE shelf. The same type of hardware can be used at both the local and remote sites with minimal card changes. The feature can also run on Televerket (TVT) or TELI hardware.

The RPE equipment consists of the following circuit packs:

— Remote Peripheral Equipment Controller (RPC)

— Path Switch (PA SW)

— Phase Locked Loop (PLL), and

— Carrier Interface (CI).

The alternative configurations of an RPE group are as follows:

— four primary loops or less, and

— three primary loops or less, plus one spare loop.

The latter configuration has greater reliability. The spare loop is automatically switched into restore service when a primary loop fails. When functioning again, the primary loop can be put into service automatically or manually in a maintenance operation.

Established voice connections are not noticeable affected by a switch to another loop. Established data calls will, in most cases, be disconnected by this disturbance. If a timeslot blocking situation is found on the loop being switched to, established calls will also be disconnected. Calls that are not in an established state when loop switching occurs will be disconnected.

Call processing will work in the same way on RPE loops as on non-RPE loops. The number of set capacity of a network loop is not affected by the introduction of 2 Mbps RPE.

The following three different transmission formats can be distinguished in a 2 Mbps RPE system:

— Standard Meridian 1 network loop transmission format. This is not suitable for CCITT PCM and must be reformatted into RPE transmission format.

— RPE transmission format. This is CCITT PCM compatible but is not optimal. Must be converted into RPE span transmission format.

— RPE span transmission format. This is a CCITT bipolar HDB3 encoded signal.

Call Reconnection

A call reconnection capability is provided on 2 Mbps RPE, which allows an established two-party call to remain established when an act of sparing or unsparing occurs. This means that the voice connection is not affected by a switch over to another loop, except for a short interval when the timeslot memory is being manipulated. Data calls, however, are usually disconnected by this disturbance. Also, any established call can be disconnected if the loop to which the switch occurs has a blocked timeslot. Calls that are not established during sparing or unsparing are disconnected.

Fault Monitoring

A fault monitoring capability detects any fault conditions pertaining to the PCM communications and the functioning of the RPE equipment, and any customer-defined condition. Failure reports are printed out on the maintenance terminal. In response to certain fault conditions, the Meridian 1 system attempts to switch over to the backup loop, if one exists. To prevent temporary failures from causing this action, tolerance level thresholds can be defined.

Alarm Handling

The alarm handling function checks for primary loop failure (several failures during a certain period, or a single failure lasting too long). If any failure is detected, automatic switching, or "sparing", to a spare loop is performed. The feature enhancement adds flexibility to how this sparing is controlled. The Counter and Timer thresholds have been changed from one set per Private Branch Exchange (PABX) to one set per RPE group, and the error counting and counter reset changed from every 24 hours to every half hour. Also, two additional maintenance information fields are printed if automatic sparing has occurred.

Two different alarm timers are used to detect cases where reported alarms remain longer than two reporting intervals. If a clearing signal is received before the first timer expires, the alarm timing is canceled and no more action taken. If the first timer expires, the second timer is started, and the "no new data calls" (NND) flag is set and the NND timer is started for the RPE loop. During this customer-defined period, no new data calls are allowed on the RPE loop. If new alarm indications are detected while a loop is in NND state, the NND flag is reset and the and timer restarted.

Refer to the *2 Mbps Remote Peripheral Equipment* (553-2741-100 to 553-2741-500 and 553-2931-100 to 553-2931-500) series of Northern Telecom Publications (NTPs) for more information on RPE hardware and operations.

Operating parameters

The maximum number of RPE groups allowed per Meridian 1 is 31.

The maximum number of RPE loops allowed per Meridian 1 is 124 (31 groups x 4 loops per group).

One network loop has to be fully assigned for remote use at one site.

A combination of service to both local and remote peripheral shelves is not permitted. (At the remote site, the RPE loop is capable of serving the same number of peripheral shelves as a local Meridian 1 loop.)

Only one attendant position per RPE loop can be configured for traffic capacity.

Emergency transfer units located in the Common Equipment (CE) cabinet will not work at 2 Mbps RPE remote sites.

No new data calls are allowed to and from Meridian 1 terminals as long as RPE loops are in the "No New Data" state.

The 1.5 Mbps RPE and 2 Mbps RPE systems can both operate on the same Meridian 1, but not simultaneously. The two features differ in electrical and transmission characteristics.

Feature interactions

There are no interactions with other features.

Feature packaging

2 Mbps Remote Peripheral Equipment (RPE2) package 165.

Feature implementation

Refer to the *2 Mbps Remote Peripheral Equipment Installation and acceptance procedures* 553-2931-200 for implementation procedures.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	16.87G
Networking:	No

2 Mbps Remote Peripheral Equipment Alarm Handling

This feature enhances the existing alarm handling function for 2 Mbps Remote Peripheral Equipment (RPE).

The alarm handling function checks for primary loop failure (several failures during a certain period, or a single failure lasting too long). If any failure is detected, automatic switching, or "sparing", to a spare loop is performed.

The enhancement adds flexibility to how this sparing is controlled. The Counter and Timer thresholds have been changed from one set per Public Branch Exchange (PABX) to one set per RPE group, and the error counting and counter reset changed from every 24 hours to every half-hour. Also, two additional maintenance information fields are printed if automatic sparing has occurred.

Operating parameters

The same operating parameters apply as for the 2 Mbps RPE feature.

Feature interactions

There are no interactions with other features.

Feature packaging

2 Mbps Remote Peripheral Equipment (RPE2) package 165.

Feature implementation

LD 52 – Define RPE group data and RPE system thresholds.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RPE2	2.0 Mbps group data.
GRP	1-31	RPE group number.
TASK	GMBR	Enter GMBR to perform Group Member task.
	TTHS	Enter TTHS to perform Timer Threshold task.
	CTHS	Enter CTHS to perform Counter Threshold task.
	NND	Enter NND to perform No New Data call timer task.
If the response to the TASK prompt is GMBR, enter the following Group Member data:		
ID	x...x	1-16 character alphanumeric RPE group identification number.
LM0	0-159	Loop number for member 0 in the group (the first primary loop). Precede with X to delete loop number.
LM1	0-159	Enter the loop number for member 1 in the group (the second primary loop). Precede with X to delete loop number.
LM2	0-159	Enter the loop number for member 2 in the group (the second primary loop). Precede with X to delete loop number.
LM3	0-159	Enter the loop number for member 3in the group (the second primary loop). Precede with X to delete loop number.
- SPAR	(NO) YES	Spare loop option.

If the response to the TASK prompt is TTHS, enter the following Timer Threshold data:		
LFAL	2-(10)-999	Loss of frame alignment threshold timer at a local site (in seconds).
FAEL	2-(600)-999	Frame alignment error rate threshold timer at a local site (in seconds).
PCML	2-(600)-999	PCM error rate threshold timer at a local site (in seconds).
LFAR	2-(10)-999	Loss of frame alignment threshold timer at a remote site (in seconds).
FAER	2-(10)-999	Frame alignment error rate threshold timer at a remote site (in seconds).
PCMR	2-(600)-999	Pulse Code Modulation error rate threshold timer at a remote site (in seconds).
RPF	128-(1024)-9999	Remote Processor failure threshold timer at a local site (in milliseconds).
If the response to the TASK prompt is CTHS, enter the following Counter Threshold data (the following values are in seconds):		
LFAL	0-(5)-255	Loss of frame alignment threshold counter at a local site.
FAEL	0-(5)-255	Frame alignment error rate threshold counter at a local site.
PCML	0-(5)-255	PCM error rate threshold counter at a local site.
LFAR	0-(5)-255	Loss of frame alignment threshold counter at a remote site.
FAER	0-(5)-255	Frame alignment error rate threshold counter at a remote site.
PCMR	0-(5)-255	PCM error rate threshold counter at a remote site.
RPF	0-(3)-255	Remote Processor failure.
LINT	0-(2)-255	Remote Peripheral Equipment initialization threshold counter at a local site.
BGTH	0-(3)-7	Number of allowable background processing unsparing attempts (if BGTH is set to 0, the background processing of LD 53 will be deactivated for this RPE group).

If the response to the TASK prompt is NND, enter the following (the following values are in seconds):		
ERTH	10-(14)-30	Error alignment threshold. Time after which the NND state is entered.
NND	0-(56)-1800	No New Data call time. Time is stored as nearest lower multiple of 8. If the value is set at 0, the Error Handling system will be deactivated for this RPE Group.

Feature operation

No specific operating procedures are required to use this feature.

Introduced in X11 Release:	14.46E
Networking:	No

16-Button Digitone/Multifrequency Operation

This feature allows the use of a 2500-type telephone with 16 buttons instead of 12 buttons. The extra keys provide single button access to features that would otherwise require the use of Flexible Feature Codes. The feature also provides an autodial function. With this feature, autodial is also available to 12-button Digitone/Multifrequency (DTMF) telephones equipped with a true ground (GRD) button and 2500-type telephones with switchhook flash and calibrated flash.

Not all telephones must share the same assignments. In LD 18, functions can be overlay programmed against a key for each of the three modes. A set of these key-function definitions can then be assigned to one or more telephone station groups. Up to 127 sets of key function assignments (called ABCD tables) are permitted.

The following Flexible Feature Code functions can be accessed using the new (A, B, C, D, * and #) keys while in the pre-dial mode (when the telephone is receiving dial tone):

— authorization code

— automatic set relocation

— automatic wake-up activate

— automatic wake-up deactivate

— automatic wake-up verify

— Call Detail Recording charge account

— call forward all calls activate

— call forward all calls deactivate

- call forward all calls verify
- call forward toggle
- call park access
- conference diagnostics
- deactivate RGA, LND, SNR, or CFW
- electronic lock phone
- electronic lock phone (remote)
- Group Hunting pilot DN
- Incoming Call Identification (ICI) activate
- ICI deactivate
- ICI print
- integrated message system access
- last number redial
- maintenance access
- pick up DN
- pick up group
- pick up ringing number
- radio paging initiate (parallel)
- radio paging initiate (serial)
- radio paging answer (parallel)
- ring again deactivate
- ring again verify
- room status
- speed call controller
- speed call erase
- speed call user
- store number (erase)

— store number (redial)

— store number (save)

— system speed call user

— trunk answer from any station

— terminal diagnostics

— trunk verification, and

— user status.

The following functions can be accessed using the new (A, B, C, D, * and #) keys while in the post-dial mode (when it receives special dial tone after a recall during an active call, or after a busy DN has been dialed):

— Call Detail Recording charge account

— call park

— Conference six trunk disconnect

— ICI override

— last number redial

— Malicious Call Trace

— override

— permanent hold

— radio paging initiate (parallel)

— radio paging initiate (serial)

— ring again activation

— speed call user

— store number (redial)

— store number (save), and

— system speed call user.

Operating parameters

All Digitone Receivers (DTRs) on the system must have the correct strap settings for full 16-button DTMF detection.

An ABCD table must be defined, and associated with a station group.

The customer must have the SPRE code defined, in order to activate FFC functions through the A, B, C, and D keys.

The Multi-party Operations feature must be present if control digits are to be used.

The user will need a 16-button DTMF 2500-type telephone to make full use of this feature.

The 2500-type telephone must be defined as a member of a station group with an associated ABCD table.

All the requirements for the existing system, customer and station combination must be met.

Feature interactions

China – Flexible Feature Codes - Busy Number Redial

BNR allowed can be a postdial function, and BNR denied can be a predial function. Both FFCs may be dialed normally from a 16-button DTMF telephone.

China – Flexible Feature Codes - Customer Call Forward

CCFA and CCFD are allowed as predial ABCD functions. They can also be dialed normally from 16-Button DTMF telephones.

China – Flexible Feature Codes - Outgoing Call Barring

The Outgoing Call Barring FFCs are not allowed as ABCD functions. They can be dialed normally from 16-Button DTMF telephones.

Flexible Feature Codes

The Flexible Feature Codes (FFC) package must be installed, or the FFC functions will not be available. However, control functions will still be available. An FFC table must be defined for the customer, or the FFC functions will not be available.

Group Hunt

Group Hunt Pilot DN (GRHP) function will not be supported. Group Hunting and Speed Call DN Access can be accessed via the Autodial function.

Italian Central Office Special Services

The special service FFC is not supported on the ABCD keys of 16-button DTMF sets.

Feature packaging

16-Button Digitone/Multifrequency Telephone (ABCD) package 144.

Dependency:

— Flexible Feature Codes (FFC) package 139.

Feature implementation

LD 17 – Modify the system hardware and software parameters to enable or disable the 16-Button Digitone/Multifrequency Operation feature:

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	CFN PARM	Configuration Record. Release 19 gate opener.
...		
PARM	(NO) YES	(No) Change to system parameters.
- ABCD	(NO) YES	16-Button DTMF (is not) is enabled.

LD 18 – Create or modify data for the 16-Button DTMF operation:

Prompt	Response	Description
...		
TYPE	ABCD	16-Button DTMF data.

Feature operation

Each button (A, B, C, D, * and #) can have up to three functions assigned to it. The function accessed when a key is pressed is determined by the mode of operation (pre-dial, post-dial or control mode). Functions are assigned to keys by way of overlay programs. The functions can be either Flexible Feature Code functions or the autodial function. An autodial number (of up to 23 digits) can be assigned to any of these buttons for either the pre-dial or post-dial modes. In addition, an autodial number can be assigned to the recall (RCAL) button in the pre-dial mode.

Introduced in X11 Release:	All
Networking:	No

500 Telephone Features

This feature allows 500-type (rotary dial) telephones to use Call Forward, Speed Call, and Permanent Hold. Since 500-type telephones do not have an octothorpe (#), the following features are activated by dialing SPRE and a two-digit access code.

— System Speed Call	SPRE + 73
— Call Forward All Calls	SPRE + 74
— Speed Call Controller	SPRE + 75
— Speed Call User	SPRE + 76
— Permanent Hold	SPRE + 77

Operating parameters

Allow or deny these features in LD 10.

Except for the SPRE codes used, feature operation is the same as with Meridian 1 proprietary telephones.

Feature interactions

2500 Telephone Features

When Special Service for 2500 Sets (SS25) package 18 is equipped, 2500 telephones also access the above listed features by dialing the SPRE and a two-digit access code.

Feature packaging

500 Set Dial Access to Features (SS5) package 73 requires Special Service for 2500 Sets (SS25) package 18.

Feature implementation

LD 10 – Enable 500 Telephone Features.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	(XFD) XFA	(Deny) allow transfer.
FTR	CFW xx	Call Forward All Calls and DN length (4-23). Enter X CFW to remove.
	SCC xxxx	Speed Call Controller and list number. Enter X SCC to remove.
	SCU xxxx	Speed Call User and list number. Enter X SCU to remove.
	SSU xxxx	System Speed Call User and list number. Enter X SSU to remove.
	PHD	Allow Permanent Hold. Enter X PHD to remove.

Feature operation

Call Forward All Calls

To forward your calls, follow these steps:

1. Lift the handset and dial SPRE + 74. You hear dial tone.
2. Dial the DN to where you want your calls forwarded.
3. Hang up.

To cancel forwarding, follow these steps:

1. Lift the handset and dial SPRE + 74. You hear dial tone.
2. Hang up.

Speed Call Controller

To update a predefined Speed Call list, follow these steps:

1. Lift the handset and dial SPRE + 75. You hear dial tone.
2. Dial the Speed Call code (0-999), followed by the telephone number it represents. If the entry is accepted, you hear silence. If the entry is not accepted, you hear fast busy tone.
3. Hang up.

To change a number associated with a list, follow these steps:

1. Lift the handset and dial SPRE + 75. You hear dial tone.
2. Dial the Speed Call code (0-999), followed by the new telephone number. The new number automatically replaces the old one. If the entry is accepted, you hear silence. If the entry is not accepted, you hear fast busy tone.
3. Hang up.

To remove an entry in a Speed Call list, follow these steps:

1. Lift the handset and dial SPRE + 75. You hear dial tone.
2. Dial the Speed Call code (0-999) you want to remove.
3. Hang up.

Speed Call User

To make a Speed Call, follow these steps:

1 Lift the handset and dial SPRE + 76. You hear dial tone.

2 Dial the Speed Call code (0-999).

3 The number is dialed automatically.

System Speed Call User

To make a System Speed Call, follow these steps:

1 Lift the handset and dial SPRE + 73. You hear dial tone.

2 Dial the System Speed Call code (0-999).

3 The number is dialed automatically.

Permanent Hold

To activate Permanent Hold while active on a call, follow these steps:

1 Flash the switchhook. You hear dial tone.

2 Dial SPRE + 77.

3 Hang up.

The call remains on hold until you lift the handset again or the other party disconnects.

Introduced in X11 Release:	19
Networking:	No

500/2500 Line Disconnect

500/2500 Line Disconnect is invoked when the Meridian 1 system detects on-hook/disconnect supervision from a party connected to a 500/2500 type port. Dial tone is sent to this port for a specified period of time (the default is six seconds) which is defined in LD 15 at the Line Disconnect Tone Timer (LDTT) prompt. Refer to the feature implementation for a list of LD 15 prompts.

It is used when the 500/2500 type port is connected to an automated attendant or voice mail. It allows the Meridian 1 system to know that it is not connected to a telephone, and to disconnect if the other telephone has hung up (e.g., during an automated message or a voice mail message).

Figure 93 illustrates how an incoming trunk call or internal call functions with 500/2500 Type Line Disconnect.

This illustration shows the incoming trunk call or internal call disconnected and dial tone being provided by the 500/2500 type port with the new Class of Service (CLS) Line Disconnect Tone Allowed (LDTA).

Figure 93
Incoming Trunk Call of Internal Call Disconnects

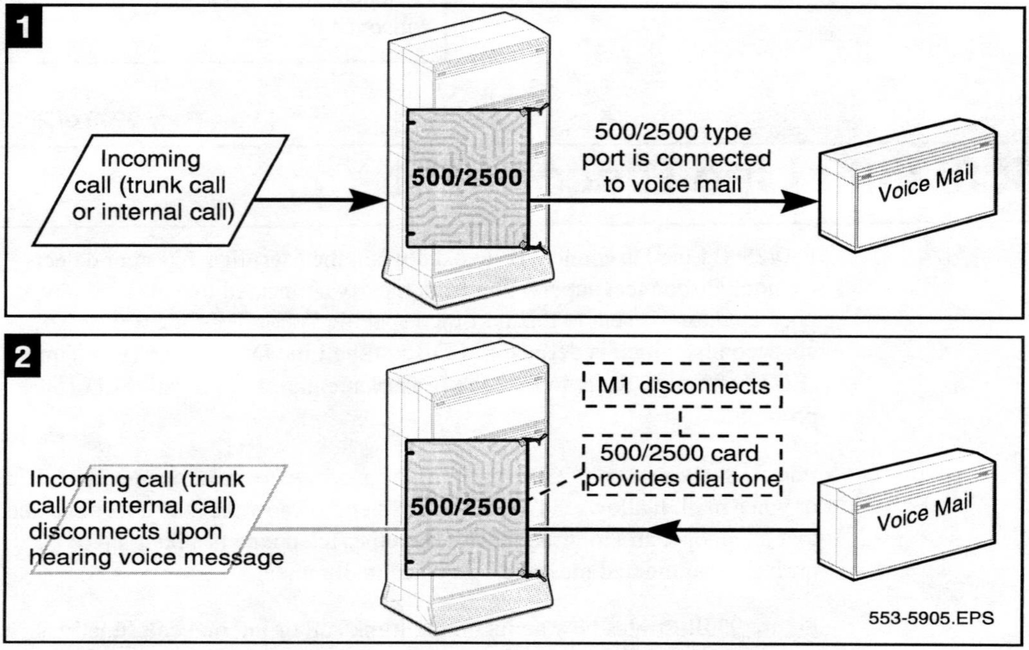
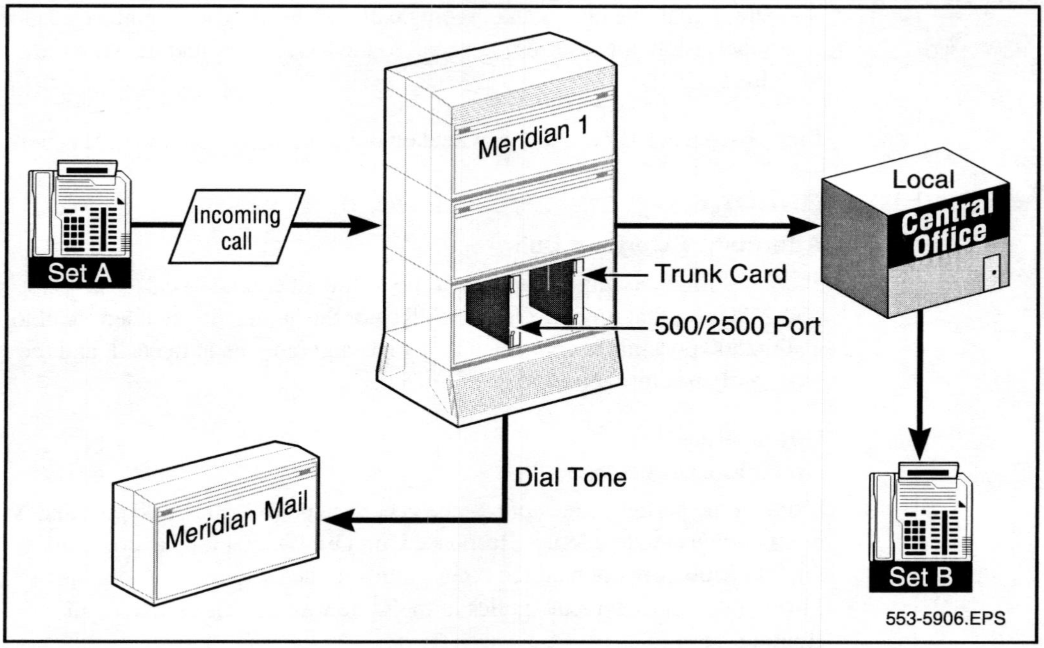


Figure 94 illustrates how an outgoing call functions with this feature. This illustration shows an outgoing call from the Meridian 1 system to the Central Office. Station A transfers Station B to Meridian Mail and goes on-hook. When Station B disconnects, dial tone is provided by the 500/2500 type port with the new Class of Service LDTA.

Figure 94
Outgoing Call from the Meridian 1 to a Central Office



Operating parameters

A 500/2500 port with LDTA Class of Service receives disconnect tone in the following cases:

— an incoming internal call is placed to an LDTA port and then disconnects

— incoming call from a trunk with disconnect supervision is placed to an LDTA port and then the incoming trunk disconnects, or

— an internal DN places an outgoing call on a trunk with disconnect supervision, then transfers the call to the LDTA port and then the trunk disconnects.

Line Disconnect Tone is not provided on outgoing calls from the LDTA port.

Feature interactions

Attendant Extended Call

500/2500 Line Disconnect applies if the attendant extends a call to a 500/2500 port that is connected to a VRU; or the attendant extended a call to a 500/2500 port that is connected to a VRU and remains in the call, and the other party has disconnected.

Conference
No Hold Conference

If one of the parties in the conference is connected to a 500/2500 port that is in turn connected to a Voice Response Unit (VRU), dial tone is provided to the 500/2500 port when all the other parties in the conference disconnect. This feature enhancement applies in the same way to Call Transfer and Hunting.

500/2500 Automatic Call Distribution agent

If a call is involved with a 500/2500 Automatic Call Distribution agent that is connected to a VRU and the other party has disconnected, 500/2500 Line Disconnect applies. When the other party disconnects, the 500/2500 agent will be returned to the idle agent queue.

Feature packaging

500/2500 Line Disconnect is included in base X11 system software.

Feature implementation

LD 10 – Allow Line Disconnect Tone for 500/2500 ports.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. Terminal Number for the Option 11.
CLS	(LDTD) LDTA (WTA) WTD	(Deny) allow Line Disconnect Tone. (Allow) deny Warning Tone.

LD 15 – Specify the dial tone timer for 500/2500 ports.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	CDB TIM	Customer Data Block. Release 21 gate opener.
CUST	0-99 0-31	Customer number. For Option 11C.
...		
- LDTT	2-(6)-30	Line Disconnect Tone timer for the 500/2500 port, in seconds.

Feature operation

No specific operating procedures are required to use the 500/2500 Line Disconnect feature.

500/2500 Line Disconnect for Outgoing Calls (Release 21)

When devices such as dictation machines are connected to a 500/2500 line port they rely on detecting a tone to indicate that the far end has released. This is necessary because the line conditions on a 500/2500 circuit do not change regardless of the status of the far end.

Currently, when a Meridian 1 detects an on-hook/disconnect supervision signal from a party on a trunk that provides disconnect supervision, and the trunk is connected to a 2500/500 port with the Line Disconnect Tone Allowed (LDTA) Class of Service, dial tone is sent for the time specified in the Customer Data Block. Thus, the device physically connected to the 500/2500 port disconnects itself and the line port as well. This functionality is used in applications requiring predictive dialing; however, previously it was limited to incoming calls.

In X11 Release 21, the 500/2500 Line Disconnect for Outgoing Calls feature expands the 500/2500 Disconnect capability to encompass outgoing calls.

Operating parameters

This feature only works with internal calls or with trunks that provide disconnect supervision. If a trunk is used that does not have disconnect supervision, the Meridian 1 does not detect the far end disconnection and the release of the call is still dependent upon the internal timing of the Automated Dialing Equipment.

This feature only applies to Automated Dialing Equipment systems capable of recognizing dial tone as a disconnect signal.

When a 500/2500 port is receiving a disconnect dial tone, it is not possible to dial a number. Dial tone cannot be broken. The port has to be released before dialing out.

Feature interactions

Attendant Extended Call

The 500/2500 Line Disconnect for Outgoing Calls feature applies if an attendant extends a call originated from a 500/2500 line port with LDTA Class of Service to a trunk or an internal extension, and the attendant has disconnected from the call. When the far end disconnects and this is a simple call, dial tone is provided to the 500/2500 line port.

Call Forward All Calls
Call Forward No Answer
Call Forward Busy
Call Forward by Call Type

The 500/2500 Line Disconnect for Outgoing Calls feature applies if a call originated from a 500/2500 line port with LDTA Class of Service is Call Forwarded to a trunk or another internal extension.

Call Transfer

The 500/2500 Line Disconnect for Outgoing Calls feature applies if a call originating from a 500/2500 line port with LDTA Class of Service is transferred by the called party to a trunk or another internal extension.

Conference
No Hold Conference

If Automated Dialing Equipment is connected to an internal extension that uses transfer or conference to include a trunk or another internal extension in the call, dial tone will be provided to the 500/2500 port when all the other parties disconnect.

Hunting

The 500/2500 Line Disconnect for Outgoing Calls feature applies if a call originated from a 500/2500 line port with LDTA Class of Service reaches a busy set that hunts to a trunk or to another internal extension.

Tone to Last Party

With the Tone to Last Party (TLP) feature configured, tones given to sets, whether involved in an internal or external call, are defined in the Tone Tables defined for the customer. If the TLP timer in the tone table is set to zero, the feature is disabled. If the TLP timer has a value greater than zero, this feature is active for all analog (500/2500 type) telephones at the customer location. The 500/2500 Line Disconnect feature takes precedence if the Tone to Last Party feature is enabled for a customer and the 500/2500 set has LDTA Class of Service.

500/2500 Automatic Call Distribution (ACD) Agents

If an Automated Dialing Equipment (ADE)/Voice Response Unit (VRU) is involved in a call with a 500/2500 ACD agent and the party disconnects, the ADE will be provided dial tone when the last party (except for the ADE/VRU) has disconnected.

Feature packaging

500/2500 Line Disconnect for Outgoing Calls feature is contained in base X11 system software.

Feature implementation

Use the same implementation procedures as for the original 500/2500 Line Disconnect feature.

Feature operation

No specific operating procedures are required to use the 500/2500 Type Line Disconnect for Outgoing Calls feature.

Introduced in X11 Release:	All
Networking:	No

2500 Telephone Features

This feature allows 2500 telephones (i.e., basic push-button sets having no feature keys) to access features otherwise available only with Meridian 1 proprietary telephones. By dialing an octothorpe (#) and a single-digit access code, 2500 telephones can access the following features:

— Call Forward All Calls Dial #1
— Speed Call Controller Dial #2
— Speed Call User Dial #3
— Permanent Hold Dial #4

Operating parameters

Allow or deny these features in LD 10.

Except for the access codes used, feature operation is the same as Meridian 1 proprietary telephones.

Feature interactions

500 Telephone Features

When 500 Set Dial Access to Features (SS5) package 73 is equipped, 2500-type telephones also access by dialing SPRE and a two-digit access code as follows:

- System Speed Call User SPRE + 73
- Call Forward All Calls SPRE + 74
- Speed Call Controller SPRE + 75
- Speed Call User SPRE + 76
- Permanent Hold SPRE + 77

Remote Call Forward

When Flexible Feature Codes (FFC) package 139 is defined and active on your system, a telephone provisioned for Call Forward in LD 10 can also Call Forward All Calls from a remote internal DN.

Feature packaging

Special Service for 2500 Sets (SS25) package 18 has no feature package dependencies.

Feature implementation

LD 10 – Enable 2500 Telephone Features.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number. For Option 11C.
CLS	(XFD) XFA	(Deny) allow transfer.
FTR	CFW xx	Call Forward All Calls and DN length (4-23). Enter X CFW to remove.
	SCC xxxx	Speed Call Controller and list number. Enter X SCC to remove.
	SCU xxxx	Speed Call User and list number. Enter X SCU to remove.
	SSU xxxx	System Speed Call User and list number. Enter X SSU to remove.
	PHD	Allow Permanent Hold. Enter X PHD to remove.

Feature operation

Call Forward All Calls

Case 1: FFC active, CFW not active

On a telephone with Flexible Feature Codes implemented, but without Call Forward currently active, use these steps to activate the feature:

1. Lift the handset and dial SPRE + 74. You hear dial tone.
2. Dial the DN where you want calls to be forwarded. The dial tone disappears.
3. Hang up to complete the activation.

To deactivate Call Forward, follow these steps:

1. Lift the handset and dial SPRE + 74. You hear dial tone.
2. Hang up to complete deactivation.

Case 2: FFC not active, CFW not active

On a telephone without Flexible Feature Codes or Call Forward currently Active, use these steps to activate the feature:

1. Lift the handset and dial #1. You hear dial tone.
2. Dial the DN where you want calls to be forwarded. The dial tone disappears.
3. Hang up to complete the activation.

To deactivate Call Forward, follow these steps:

1. Lift the handset and dial #1. You hear dial tone.
2. Hang up to complete deactivation.

Case 3: FFC active, CFW active

On a telephone with Flexible Feature Codes and Call Forward currently active, use these steps to deactivate the feature:

1. Lift the handset and dial #1. You hear confirmation tone.
2. Hang up to complete the deactivation.

To reactivate Call Forward, follow these steps:

1 Lift the handset and dial #1. You hear dial tone.

2 Dial the DN where you want calls to be forwarded. The dial tone disappears.

3 Hang up to complete the activation.

– or –

1 Lift the handset and dial #1. You hear dial tone.

2 Dial the DN where you want calls to be forwarded. The dial tone disappears.

3 Dial the EOD string. You hear a confirmation tone.

4 Hang up to complete the activation.

– or –

1 Lift the handset and dial #1. You hear dial tone.

2 Hang up to complete the activation. Calls are forwarded to the last Call Forward DN used by this telephone.

Speed Call Controller

To update a predefined Speed Call list, follow these steps:

1 Lift the handset and dial #2. You hear dial tone.

2 Dial the Speed Call code (0-999), followed by the telephone number it represents. If the entry is accepted, you hear silence. If the entry is not accepted, you hear a fast busy tone.

3 Hang up.

To change a number associated with a list, follow these steps:

1. Lift the handset and dial #2. You hear dial tone.
2. Dial the Speed Call code (0-999), followed by the new telephone number. The new number automatically replaces the old one. If the entry is accepted, you hear silence. If the entry is not accepted, you hear a fast busy tone.
3. Hang up.

To remove an entry from a Speed Call list, follow these steps:

1. Lift the handset and dial #2. You hear dial tone.
2. Dial the Speed Call code (0-999) you want to remove.
3. Hang up.

Speed Call User

To make a Speed Call, follow these steps:

1. Lift the handset and dial #3. You hear dial tone.
2. Dial the Speed Call code (0-999).
3. The number is dialed automatically.

System Speed Call User

To make a System Speed Call, follow these steps:

1. Lift the handset and dial SPRE 73. You a dial tone.
2. Dial the System Speed Call code (0-999).
3. The number is dialed automatically.

Permanent Hold

To activate Permanent Hold while on a call, follow these steps:

1. Flash the switchhook. You hear dial tone.
2. Dial #4.
3. Hang up.

The call remains on hold until you lift the handset again or the other party disconnects.

Meridian 1

X11 features and services

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